VoIP Communication

CX100 IP PBX Telephone System

User Manual

CX100

Menu

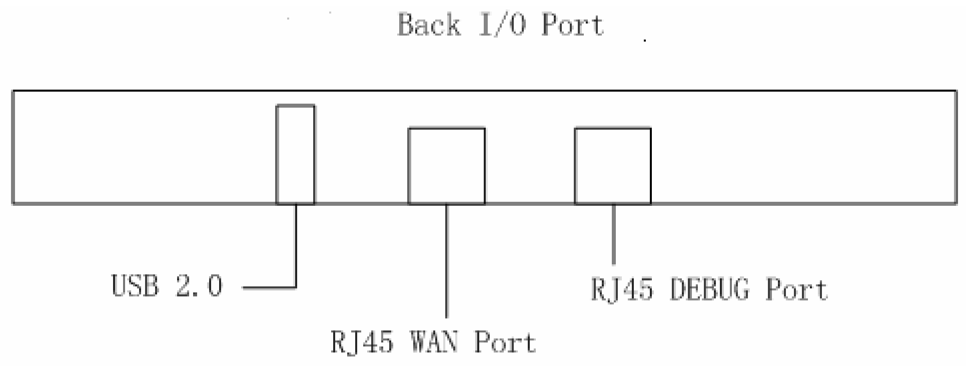
Welcome to CX100

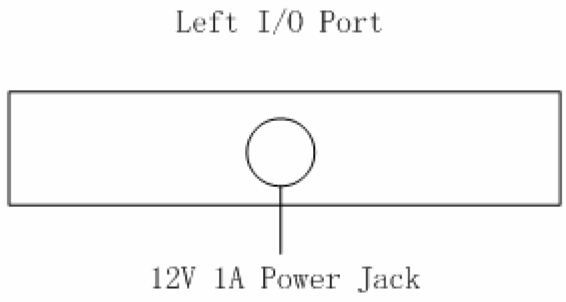
Getting Started

Thank you for purchasing CX100.It is with the excellent performance and cost, yet easy-to-configure IP PBX in the market today.
Administrating a VoIP system can be a daunting task for administrators unfamiliar with VoIP. This guide is designed to help you plan and configure CX100 Voice over IP (VoIP) deployments.

Introduction

The CX100 is the ideal system for small businesses and home offices requiring a pint-sized yet powerful on-premise IP PBX. Keeping up with the demands of sustainability, the IP PBX CX 100 is based on a low-power, high performance MIPS processor, providing the complicated communication features including the hardest HD communication protocol.
The compact solid-state device support 32 extensions and offers a wide range of IP PBX telephony features. At the same time, it is the first cheapest IP PBX in the world when it comes out.





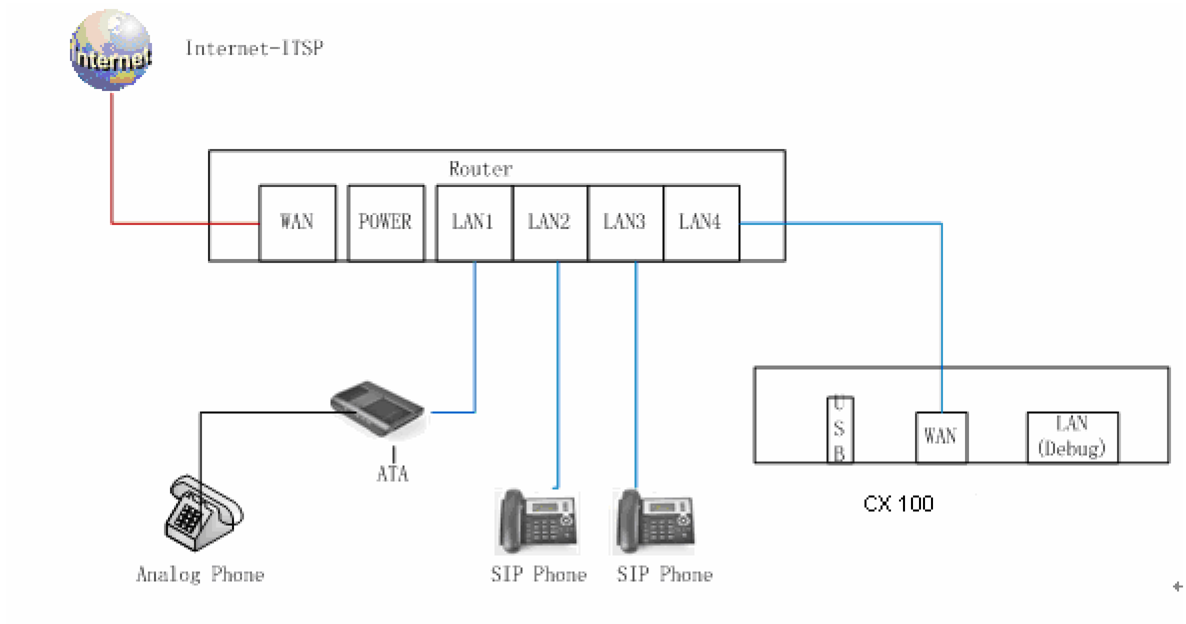
Packing list

1 unit CX100
1 Power Supply

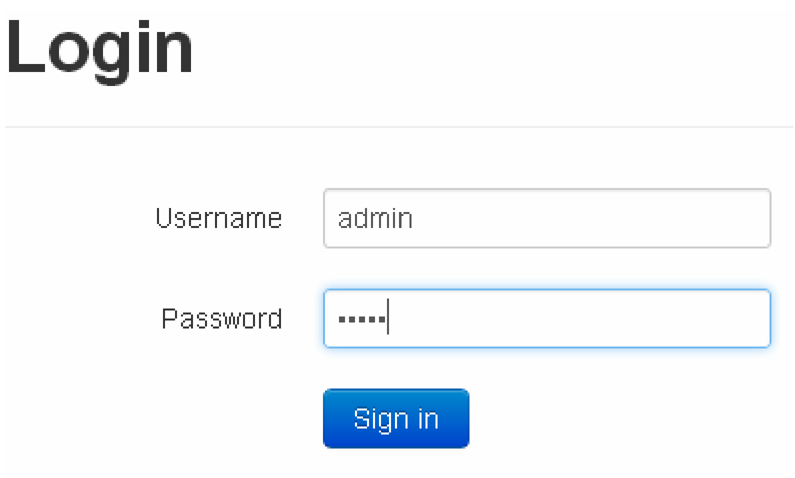
Specification

MIPS Processor
64MB RAM/16MB FLASH
1xRJ45 10/100MB Ethernet port WAN
1xRJ45 Debug port
Reset Button
Power adapter: AC 100~240V input and DC 12V/1A output
Power consumption: 1.2-2.0W
Net weight: 100g
Dimension: 150mm(L) x 90mm(W) x 31.5mm(H)
Operating humidity: 10~95%
Operating temperature: 0~45°C

Hardware setup

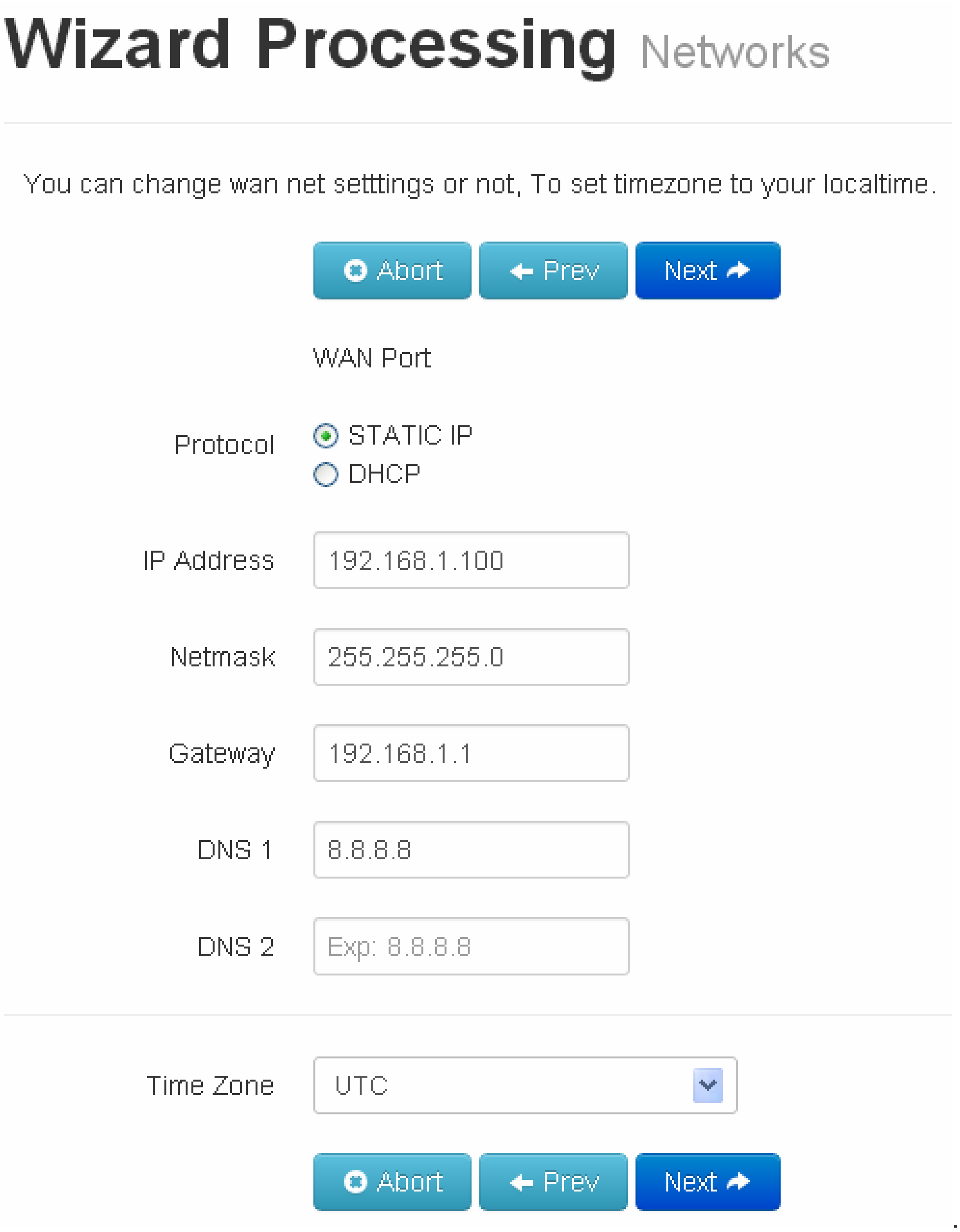


You may check the above picture to configuration.
Step 1: Connect the WAN port of CX100 with your corporate IP network. Before you connect the AX100 to the network, please check if your network can work normally.
Step 2: Plug in and open your browser to visit the address: Http://192.168.1.100:9999
Make sure your IP address is 192.168.1.XX
(If you use IE6 and above, the prefix address **Errore. Riferimento a collegamento ipertestuale non valido.** can't be left out)
Now we access to the Wizard page.
Username: admin (By default)
Password: admin (By default)



First Login to Wizard

This is your first time to log in, it will show Wizard Processing. It is simple and brief to deploy. In most cases, the default settings can be used for the rest of the configuration.



When you Abort, it will quit the Wizard. And if you want to access Wizard, just click the Wizad.

You can choose the protocol Static IP or DHCP.

Using a static IP address is the most reliable way to ensure your server IP address does not change. To find an IP address that is not in use on your network and will not be used for another client by the DHCP server or used by some other devices.

After you have found a suitable server IP address [192.168.60 is a common address], enter it and addresses for the following in the network configuration questionnaire: Netmask.
By default, the system enters 255.255.255.0. If your netmask address is different, change it to the correct value.

Gateway. The address for your gateway is usually the IP address of your router. A common gateway IP address is 192.168.0.1.

DNS server. This address will be for either a server inside your network or for a server your internet service provider (ISP) runs for your use and for which the ISP will provide you the addresses.

After all addresses have been entered and the system asks if they are correct, click Yes to accept them; otherwise, correct them, then click Yes.

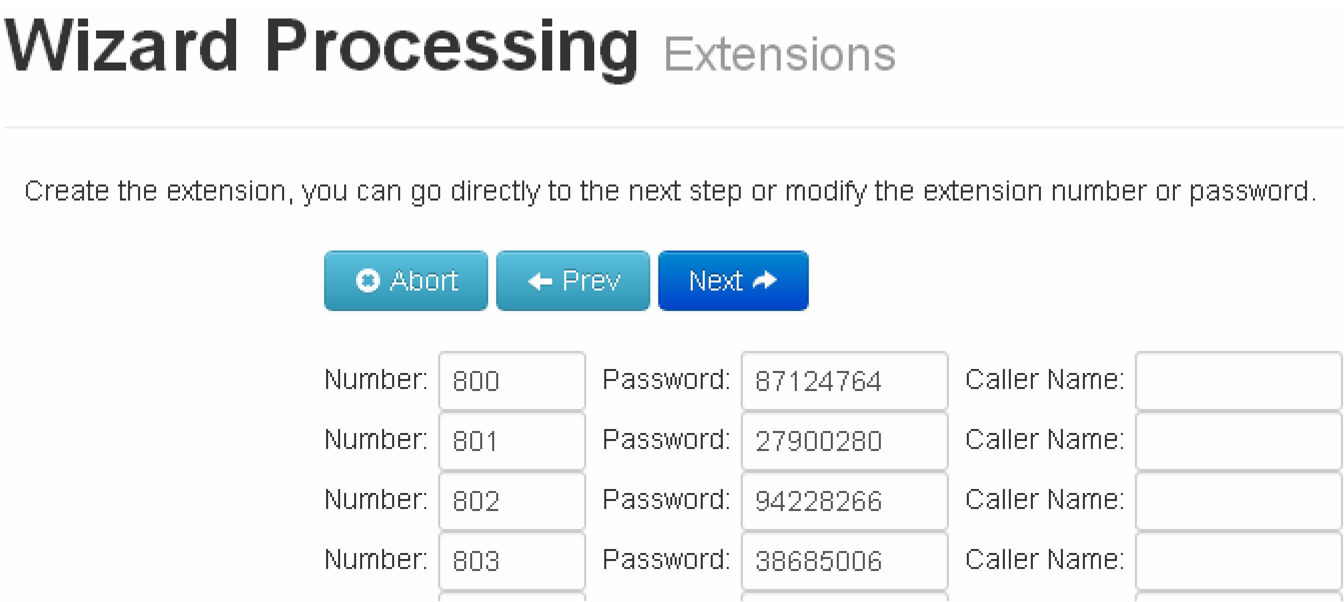
After the addresses are accepted, the system informs you that you need to reboot to use the new IP address. To reboot, click OK.

Time Zone

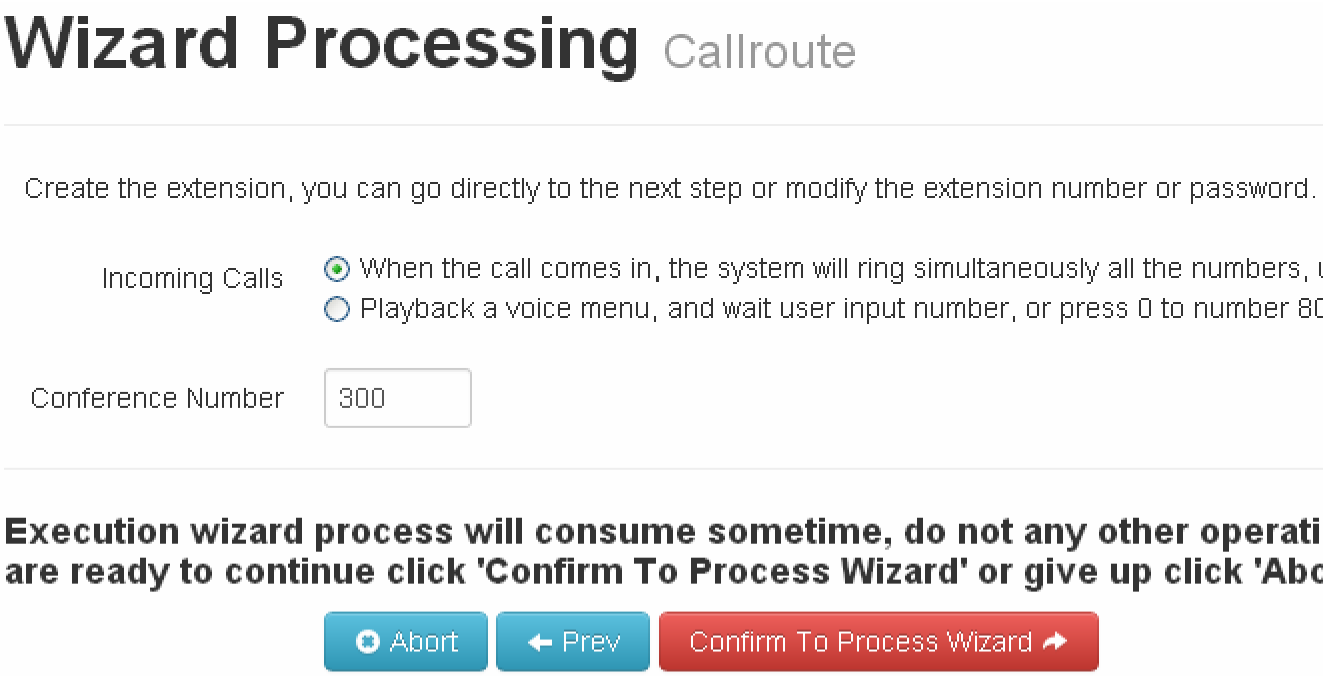
It is used to set the local time zone and is important for generating accurate call reports after your system is operating. And the features-Time Frame will also judge your working time or others according to your system time.
If you select the incorrect time zone, or you move to a different time zone later, you can change it in the Wizard.
You can follow the default setting and Next.

Now we come to the Extension page.

You can set up to 32 SIP extensions. The number length should be at least 3 digits and the password should be at least 8 digits.



Next we come to the Wizard Processing page.



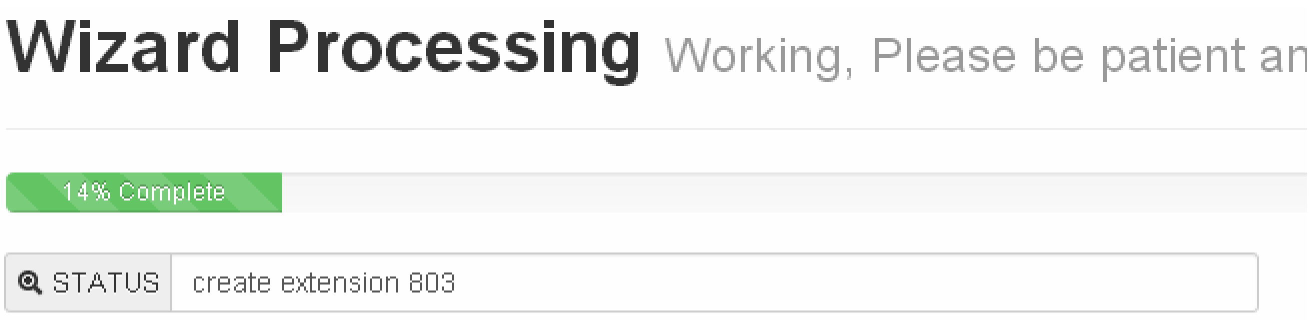
You can do nothing and click confirm.

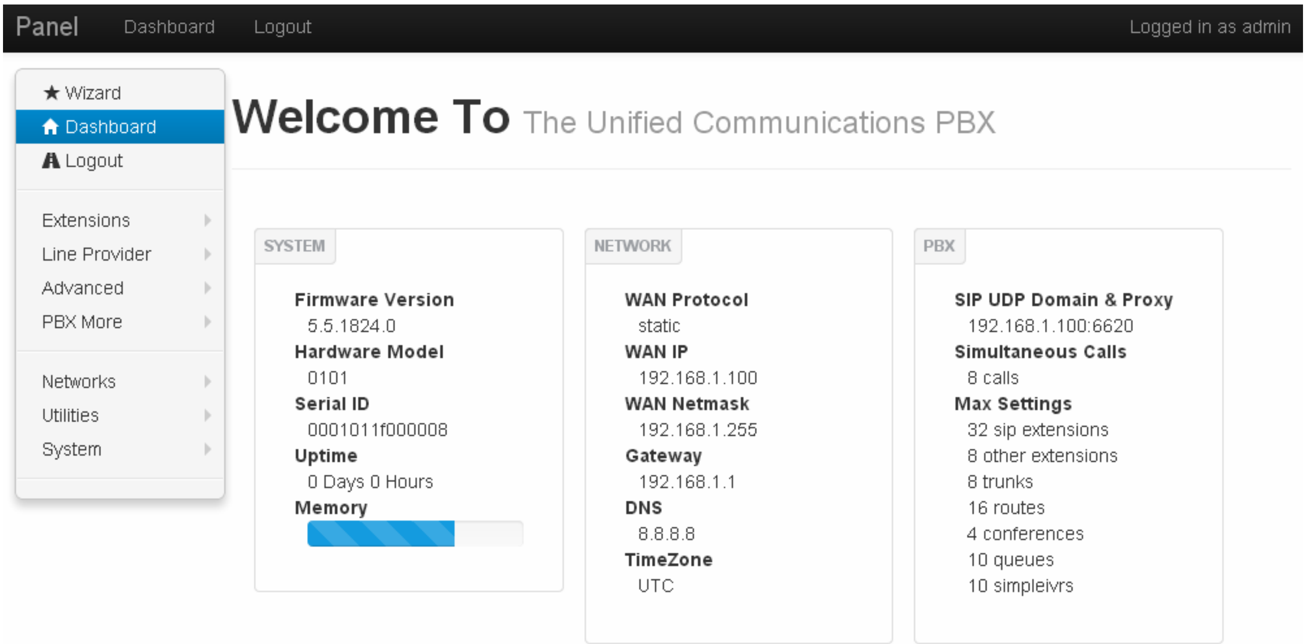
Or you can choose the two ways of incoming calls.

1. When the call comes in, all the extensions will ring and until someone to pick it up.
2. Playback a voice menu and wait the user input number or press 0 to number 800

Besides, you can also set a conference number.

Then the AX100 begin to configure. It will auto restart the PBX and if you can't see the page you can refresh it. And re-log in, you will see the dashboard of CX 100.



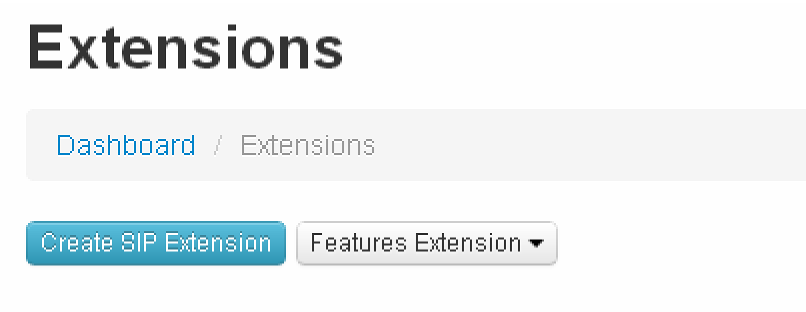


OK. The first time to access the AX100.

If you want to access to the Wizard again, just click the Wizard.

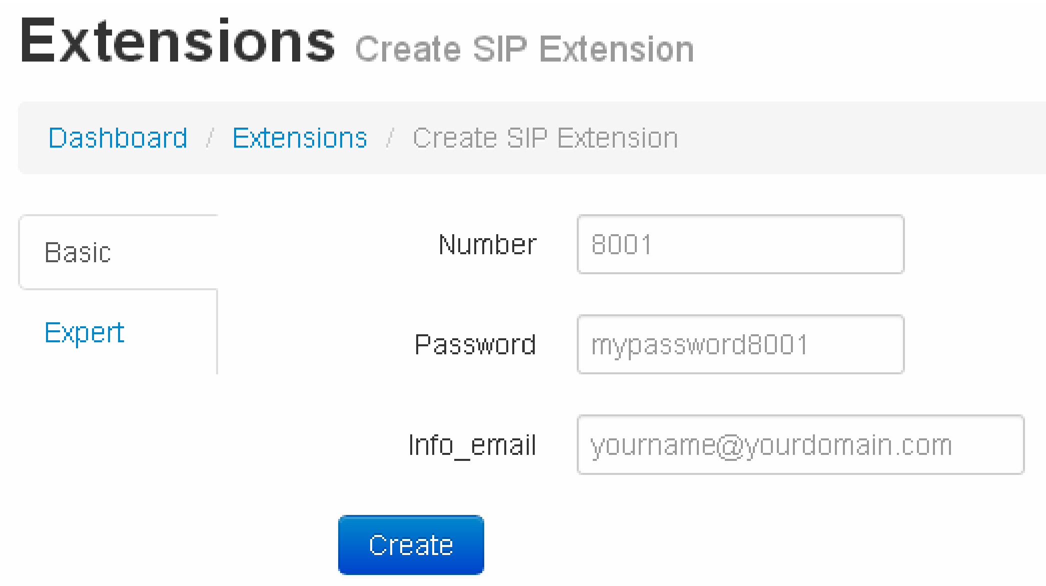
Extension

Here you can create SIP extensions and Feature Extensions.

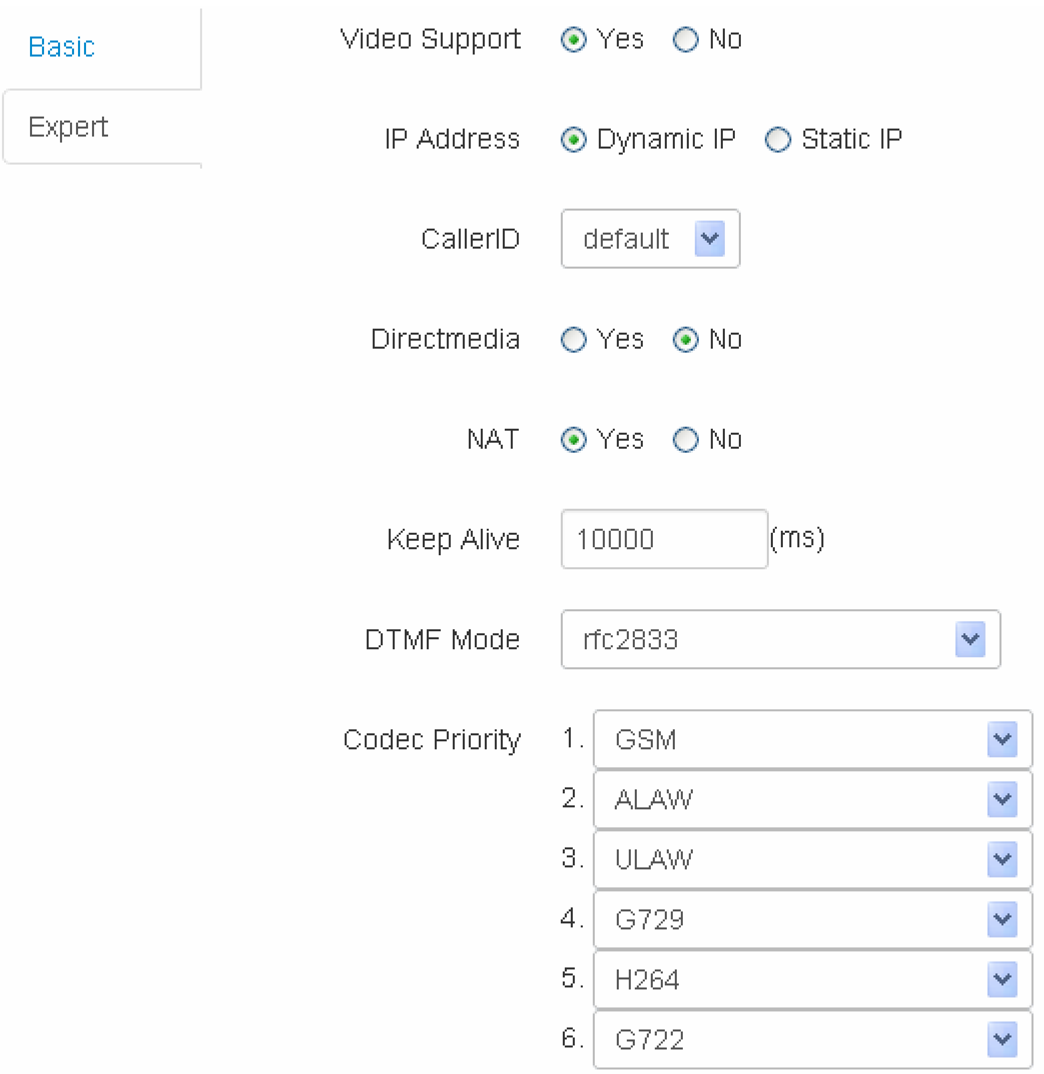


SIP Extension

SIP (Session Initiation Protocol) [RFC 3261, 3262, 3263, 3264, and 3265]
A signalling protocol for initiating and terminating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality (it is used mainly for voice and video calls over the Internet or data networks).



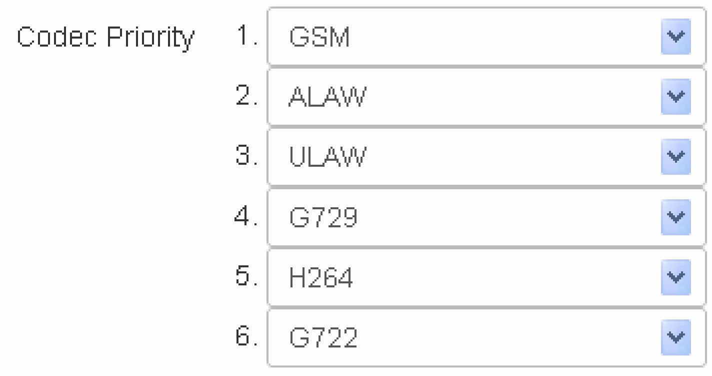
Create the SIP extensions: set the number (not less than 3 digits), password (not less than 8 digits) and email address.
Info_email: you can also set the email address.
After you set and add one account. Remember the SIP account and password and it will help to configuration your SIP Phone.



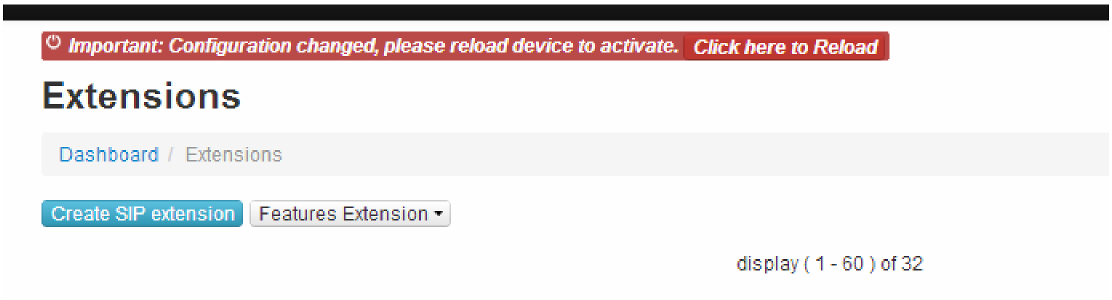
For the Expert use, you can set the Video Support, IP address and so on
Video Support: Choose Yes or No.
IP address: Dynamic IP or Static IP. If you choose the Static IP, just fill your IP address and make sure it is the same with this device.
Call ID: You can use the extension number (Default) or use this number you set yourself.

Direct Media: The default is No.
If you choose yes, it will not transfer the voice data.
Nat: Please see the glossary
Keep Alive: Keep in touch with the contacted device
DTMF Mode refers to the types of tones a phone can send and receive. Refer to your phone's user manual to find the type of DTMF tones used by your particular phone. Unless you are certain this setting needs to be changed, leave it at the default value, rfc2833.

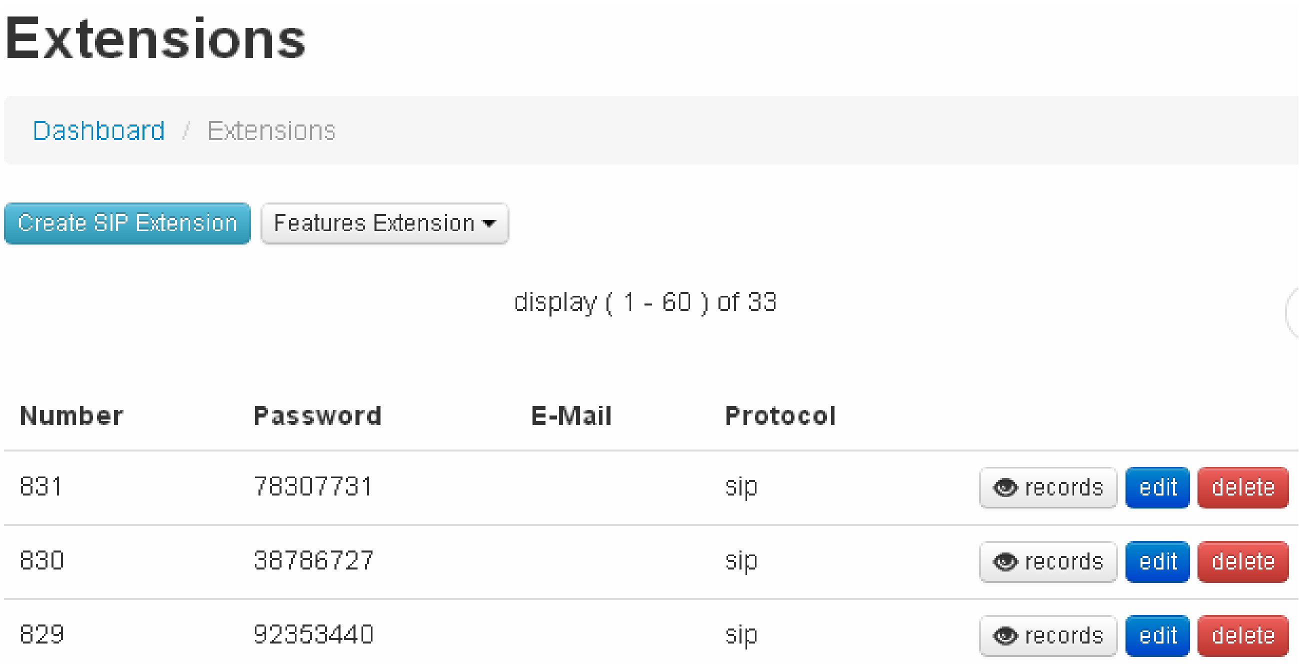
Codec Priority: You can choose the codec priority here.



After you finished, just click save and you will see the red bar to remind you to reload. Click it and wait to reload.

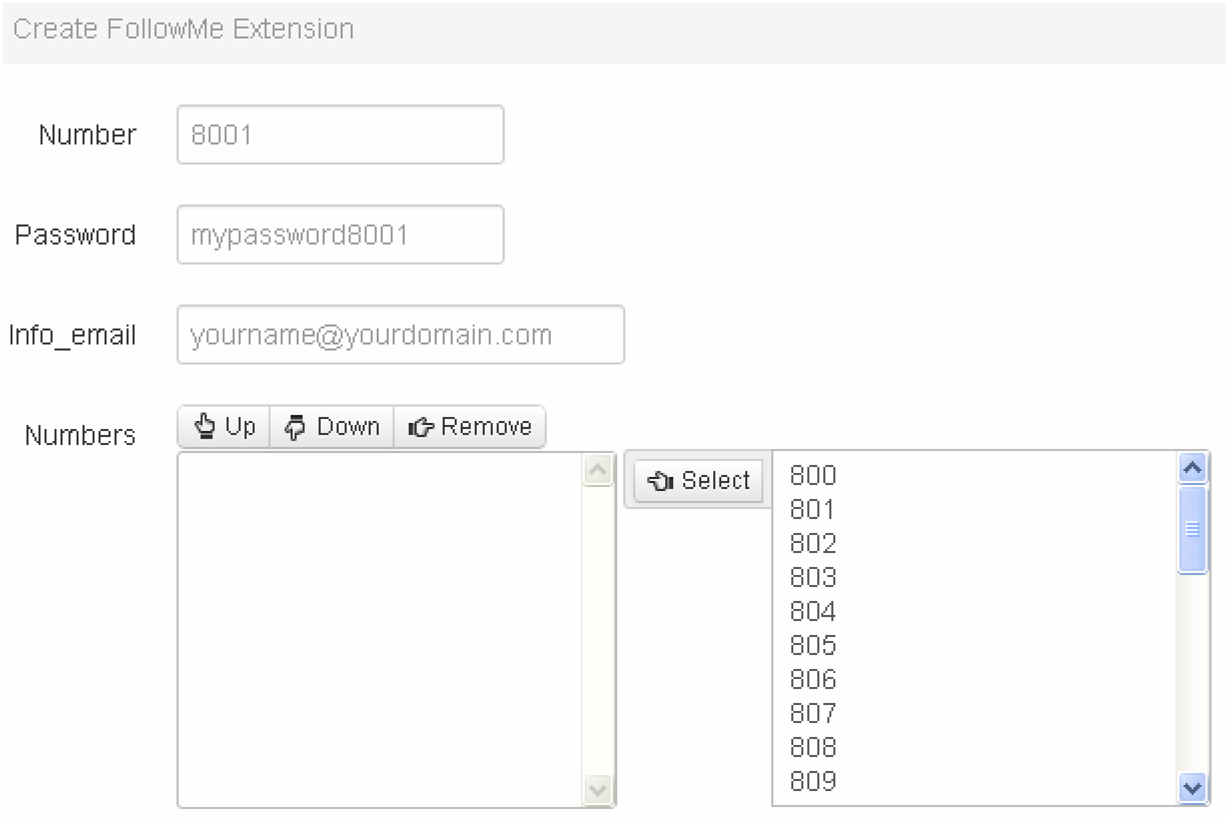


To make changes to the way an extension is set up, click on its edit button and edit the settings on the settings detail page. To remove an extension from your system permanently, click on its Delete button.



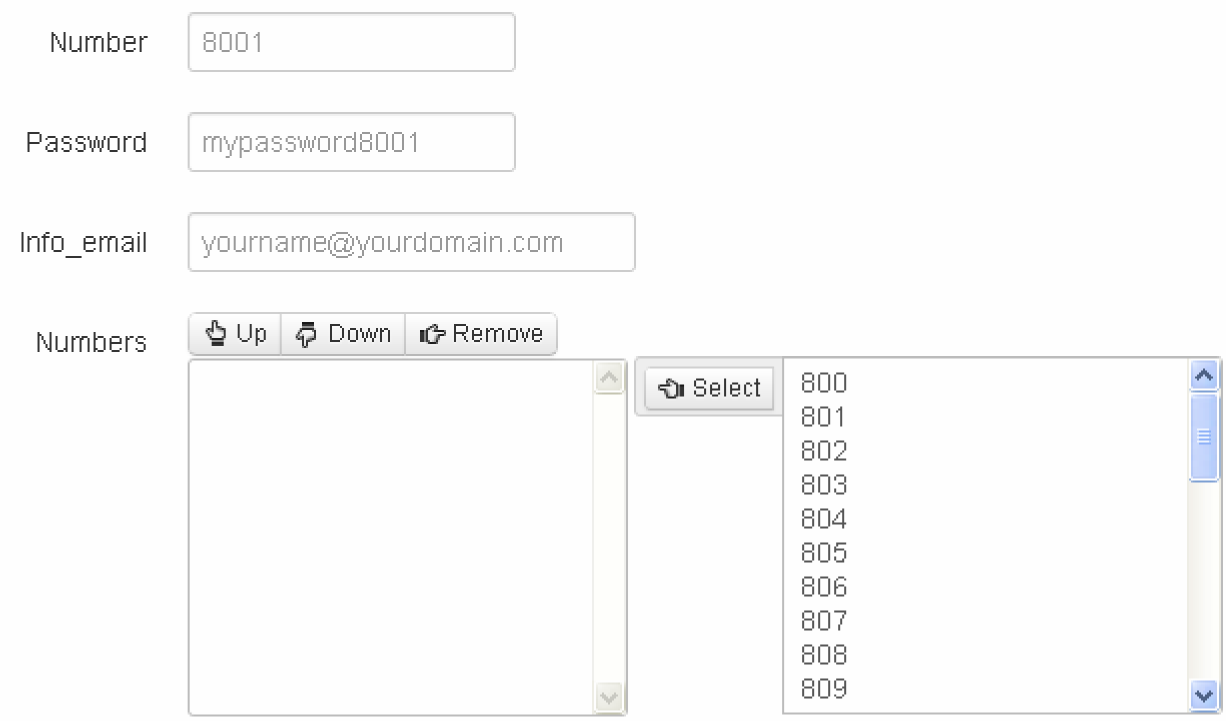
Follow Me

One of the features extensions is Follow Me.
If you have more than one number, when a call comes in, the system will ring your numbers one by one until someone answered.
You can set the detail in basic and expert. The expert is not available now.
You can set the order of the extensions number to ring.



Ring Group

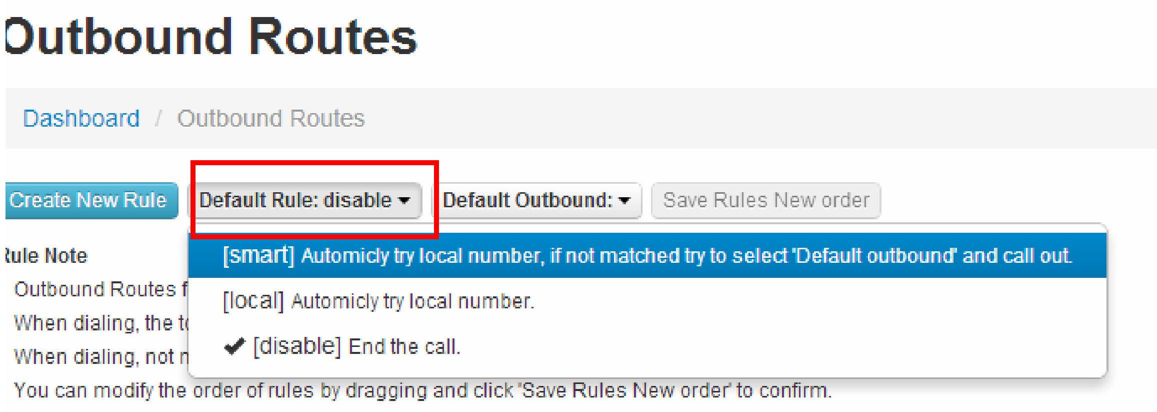
When the call comes in, all the extensions will ring simultaneously until someone to pick it up.



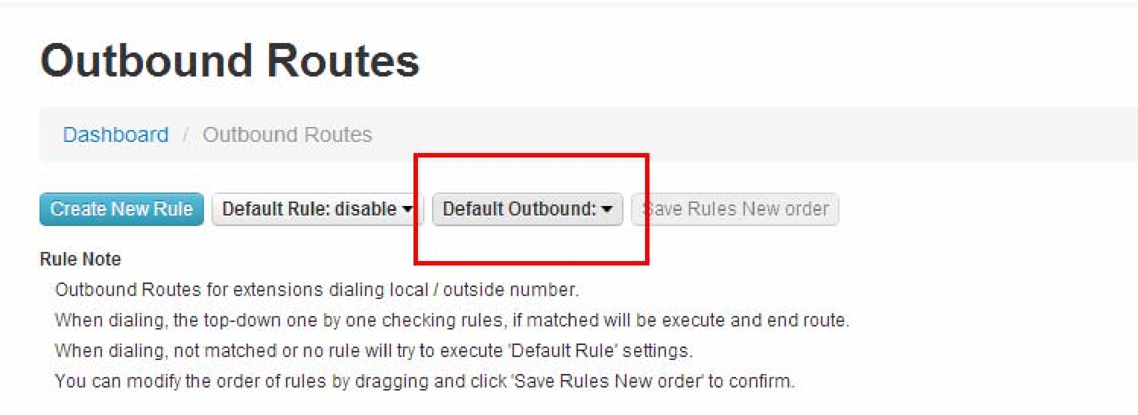
Outbound Routes

Outbound Routes for extensions dialing local / outside number.
When dialing, the top-down one by one checking rules, if matched will be execute and end route.
When dialing, not matched or no rule will try to execute 'Default Rule' settings.
You can choose the default rule. In most cases, the default settings can be used for the rest of the configuration.

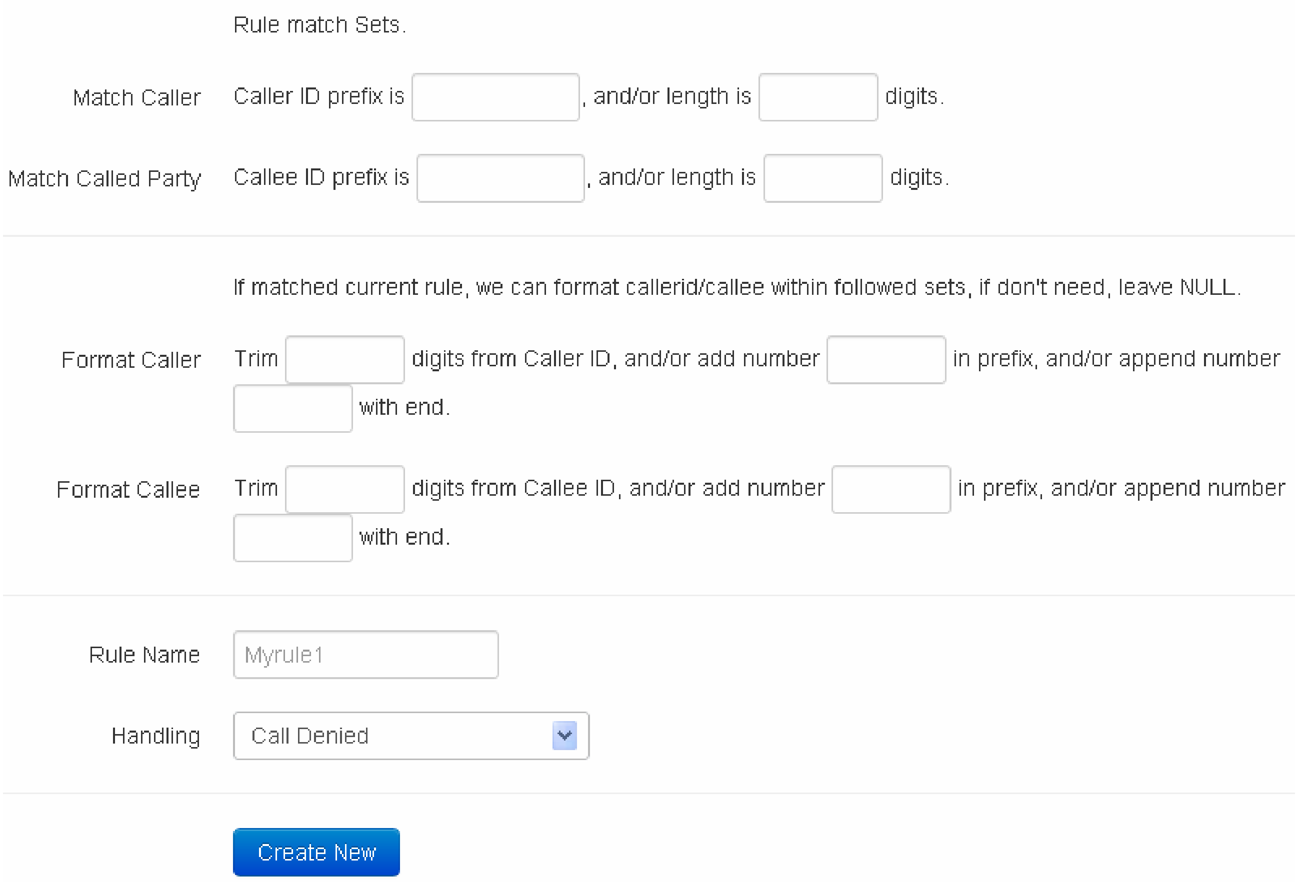
Default Rules:
[Smart]Automatically try local number, if not matched then try to select [Default Outbound] and call out.
[Local] Automatically try local number
[Disable]End the call



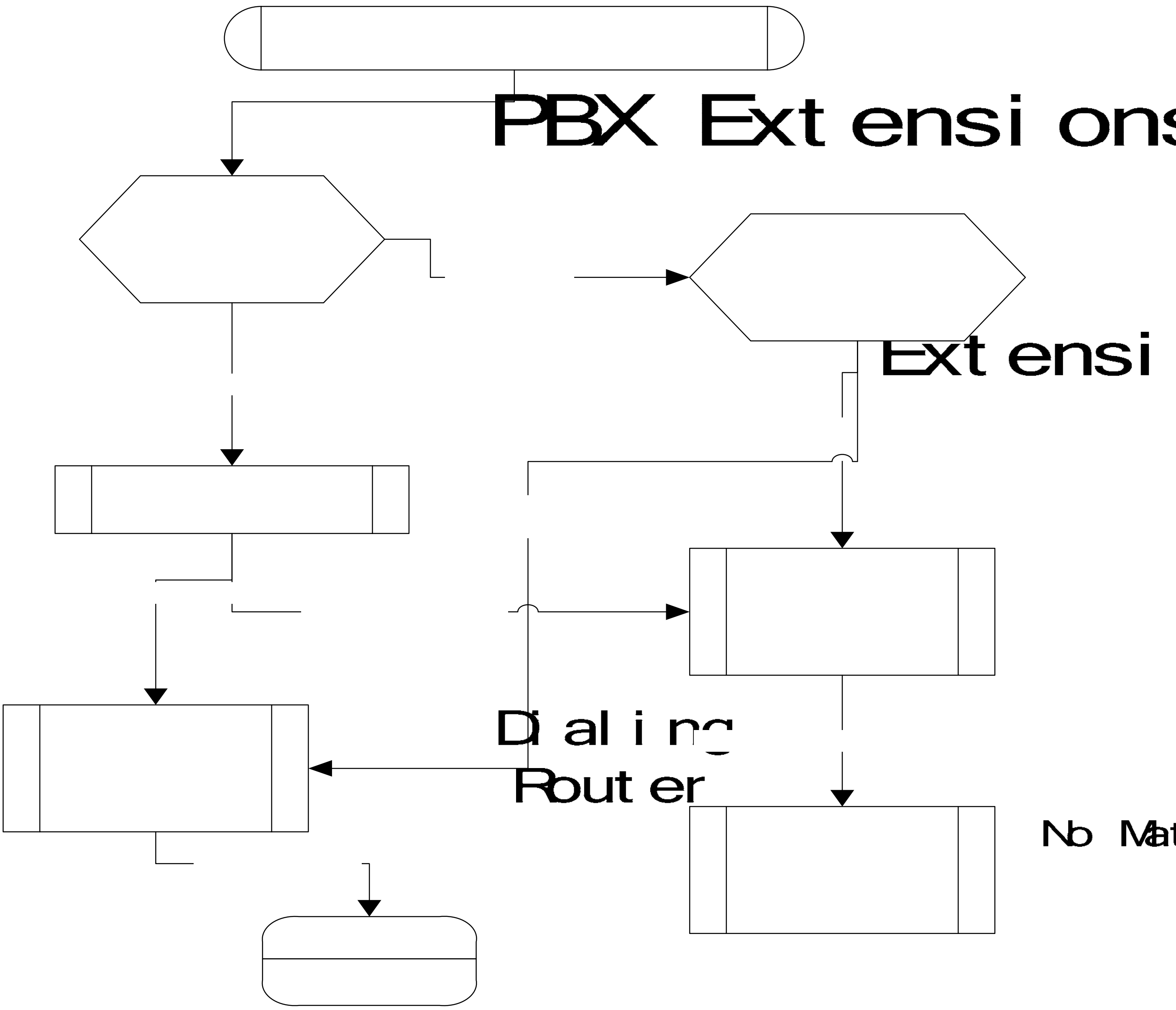
Default Outbound : You can choose the default outbound line here.



Create New Rule first:
Now you can set the outbound routes.
You can refer the tips.Set the rules to match caller ID or match the called party ID (Callee), then set the relevant executive rules: we can format caller Id/ callee ID .If don't need, leave NULL.



Below is the extension outbound routes diagram.



Matched a rule

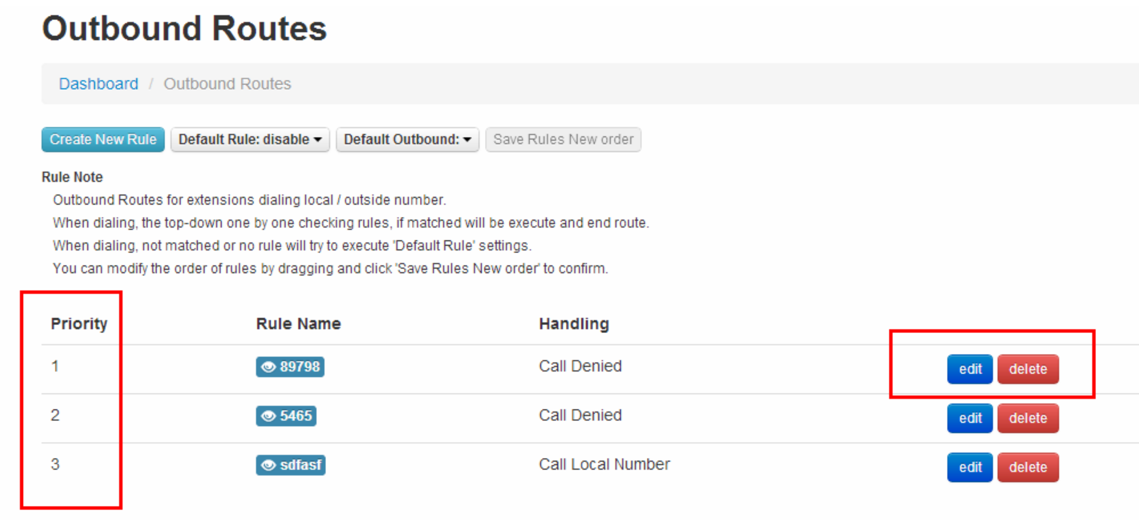
After you create them, you can modify the order of rules by dragging and click 'Save Rules New order' to confirm. And you can also edit and delete the rules.

Rule Format
Caller/Callee

Defaul
Try o

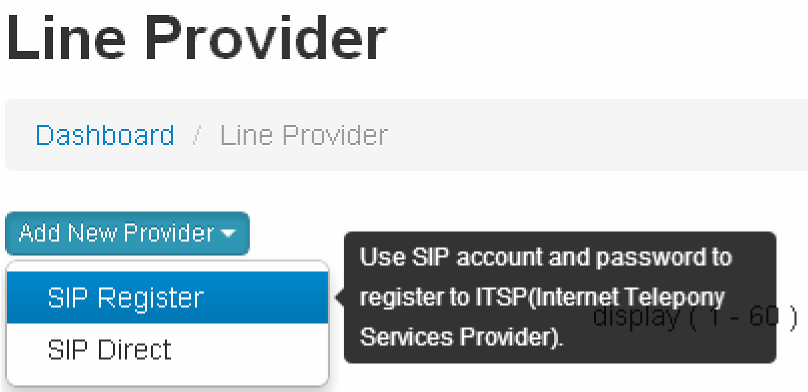
Try outbound

Try Local number



Line Provider

You can set the line provider here. Sip Register and Sip Direct.

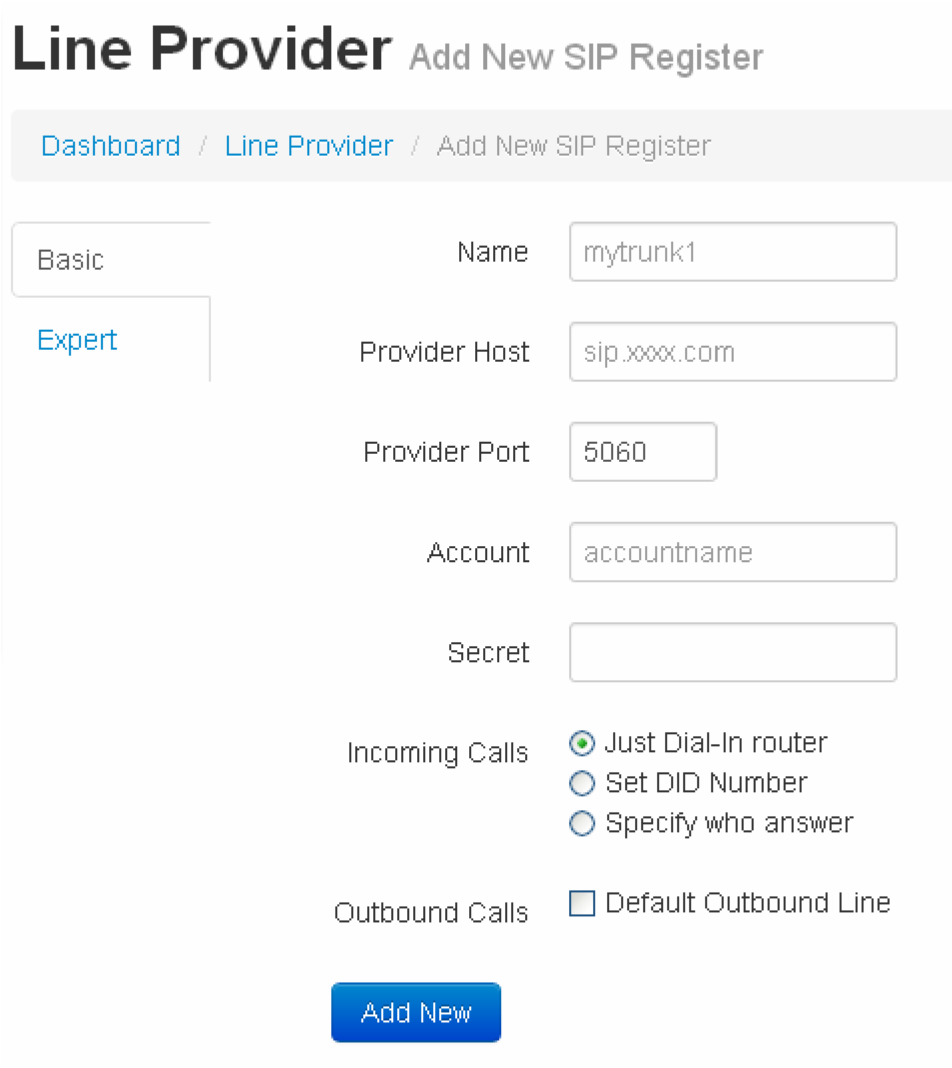


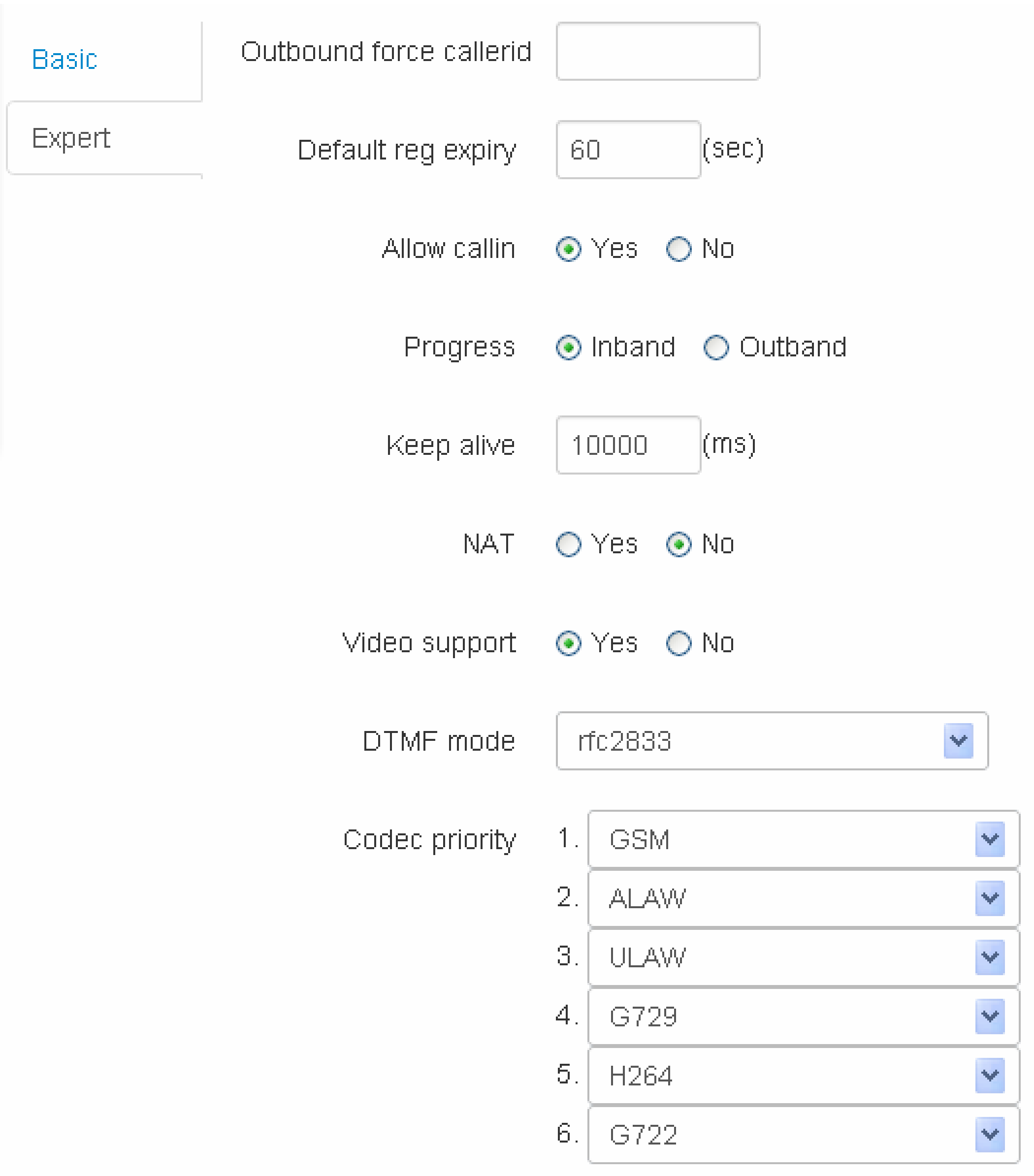
After you finished the register information, remember to tick the Outbound Calls-Default outbound calls

.

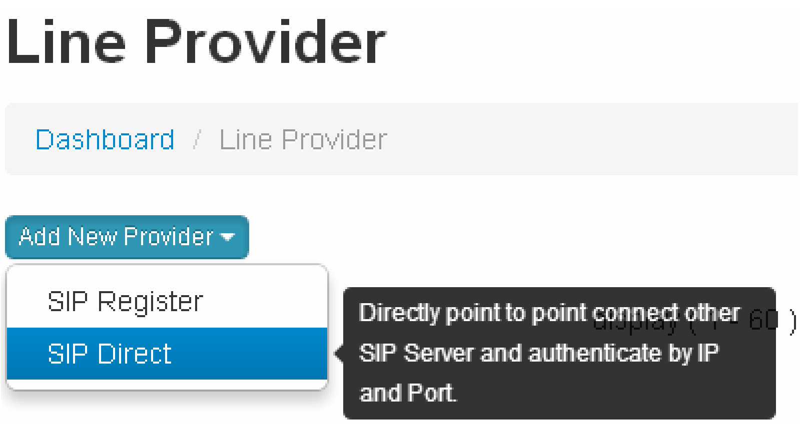
SIP Register

SIP Register: Use a SIP account and password to register to the ITSP.

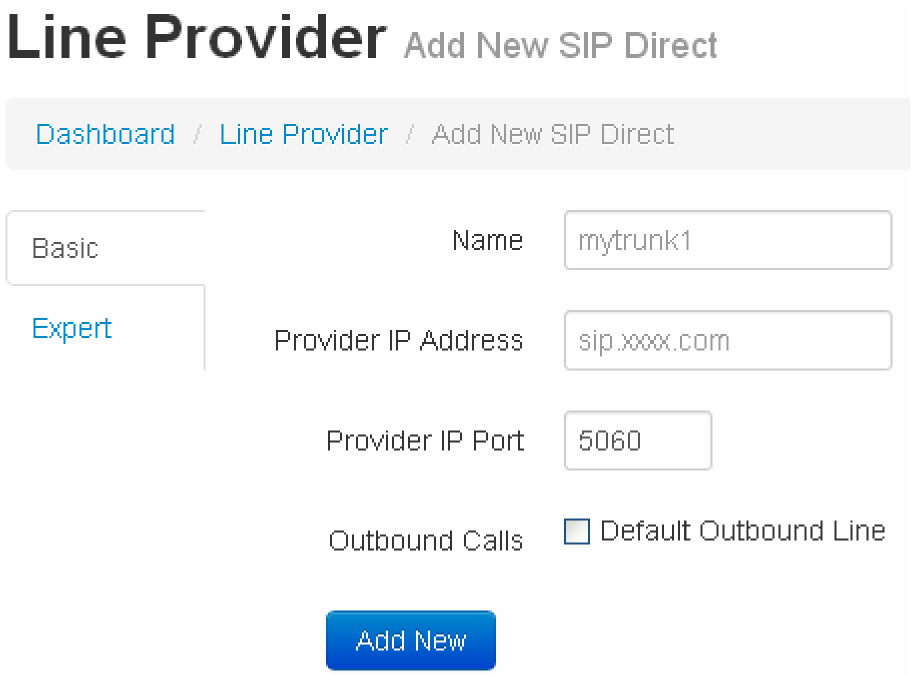

Line Provider Add New SIP Register
Dashboard / Line Provider / Add New SIP Register
Basic
Expert
Name
mytrunk1
Provider Host
sip.xxxx.com
Provider Port
5060
Account
accountname
Secret
Incoming Calls
Just Dial-In router
Set DID Number
Specify who answer
Outbound Calls
Default Outbound Line
Add New



SIP Direct

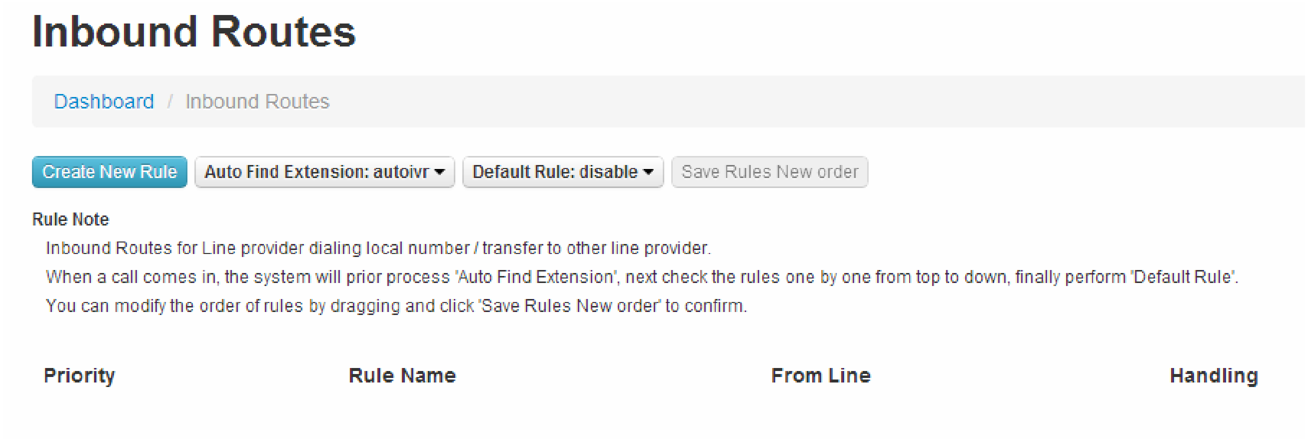


You can create the SIP direct.

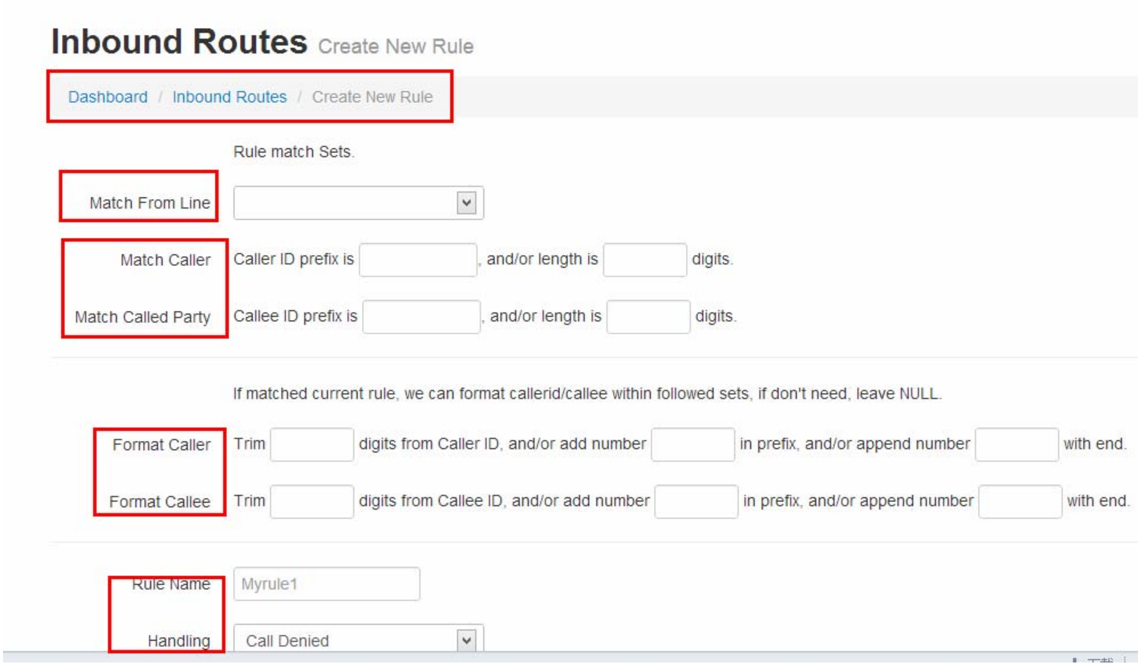


Inbound Routes

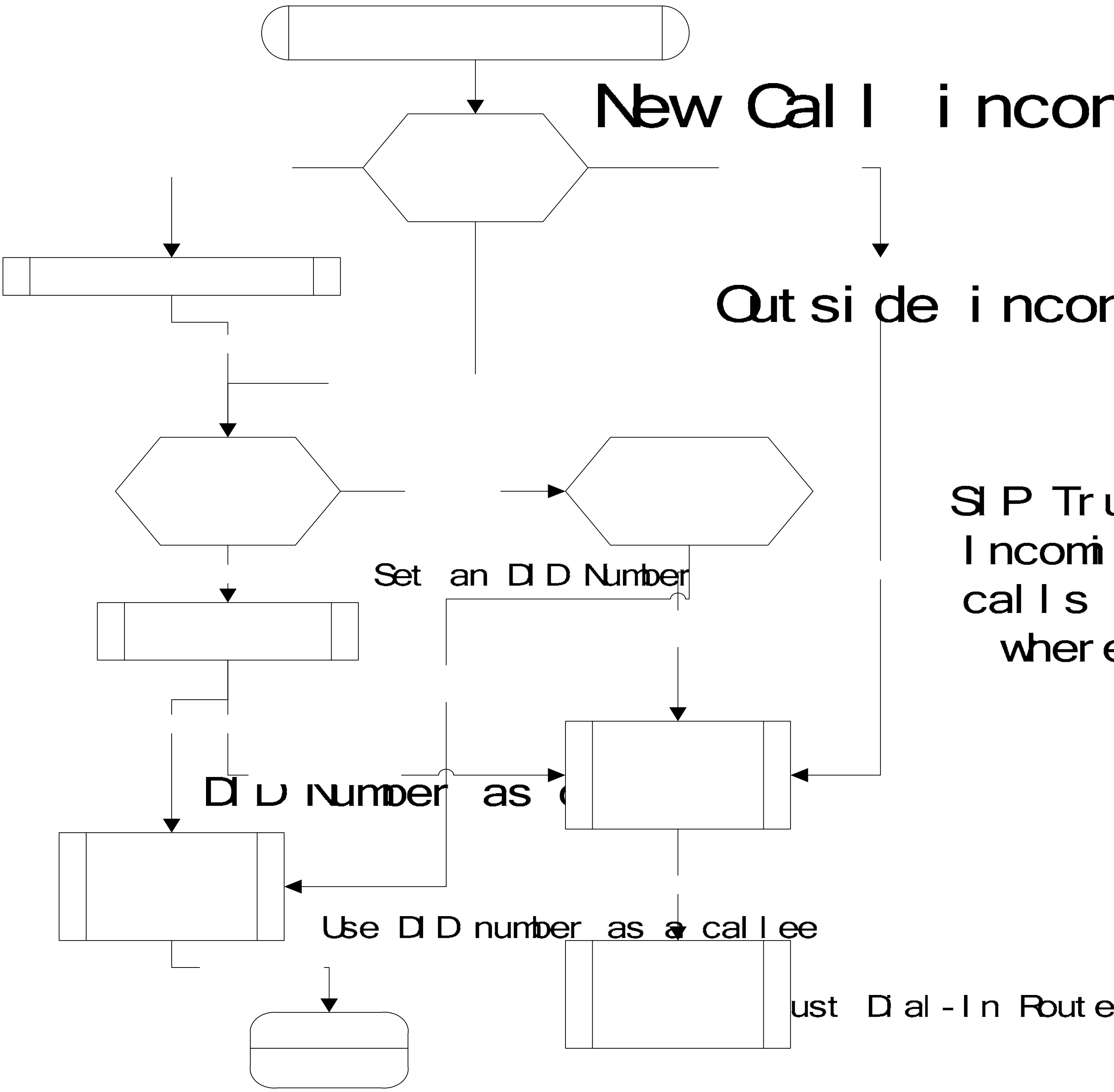
Inbound Routes for Line provider dialing local number / transfer to other line provider.



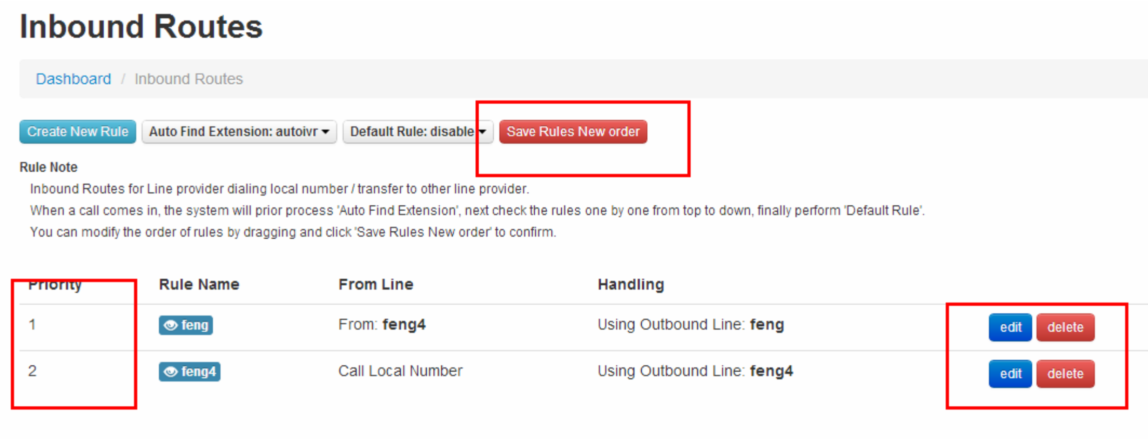
Create a new rule as below, you can create the new rules for the incoming calls.
:
Note: The default settings can be used for the rest of the configuration.



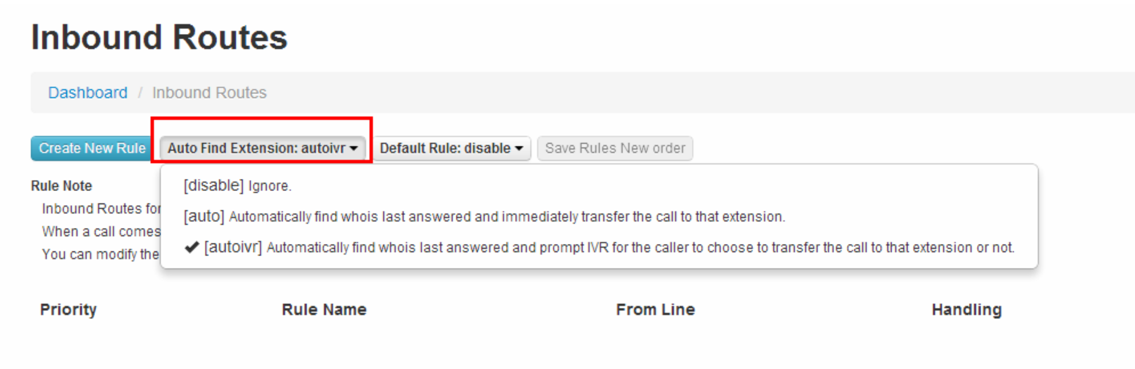
The rule setting is similar to outbound routes. You can refer the tips. Below is the diagram.

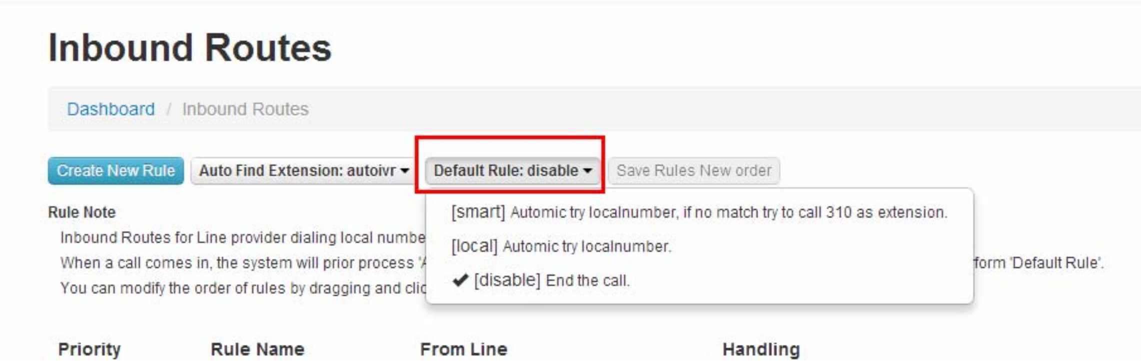


After you set several rules, you can edit or delete them.
You can also modify the order of rules by dragging and click 'Save Rules New order' to confirm.



You can modify the order of rules by dragging and click 'Save Rules New order' to confirm. When a call comes in, the system will prior process 'Auto Find Extension', next check the rules one by one from top to down, finally perform 'Default Rule'.



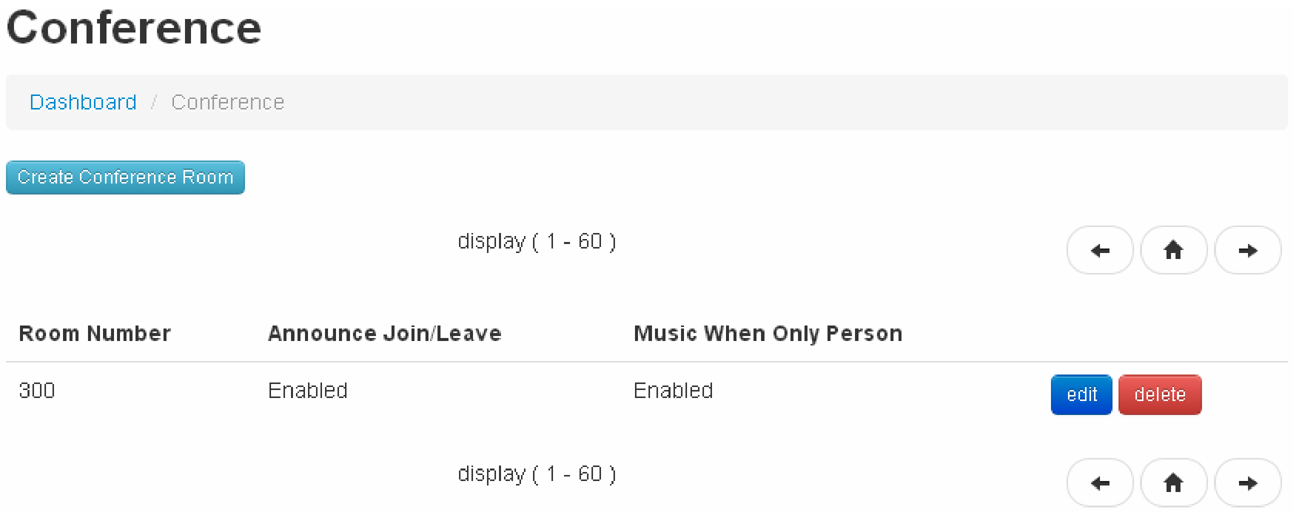


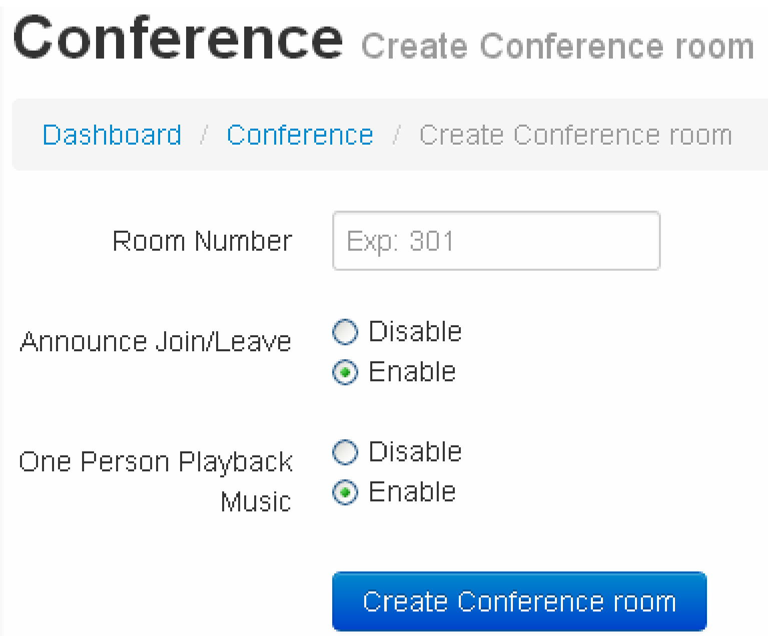
Advanced

In this section, you can set the Conference, Queues, Simple IVR and Time-Frames.

Conference

You can set a Conference here. It is simple and easy.

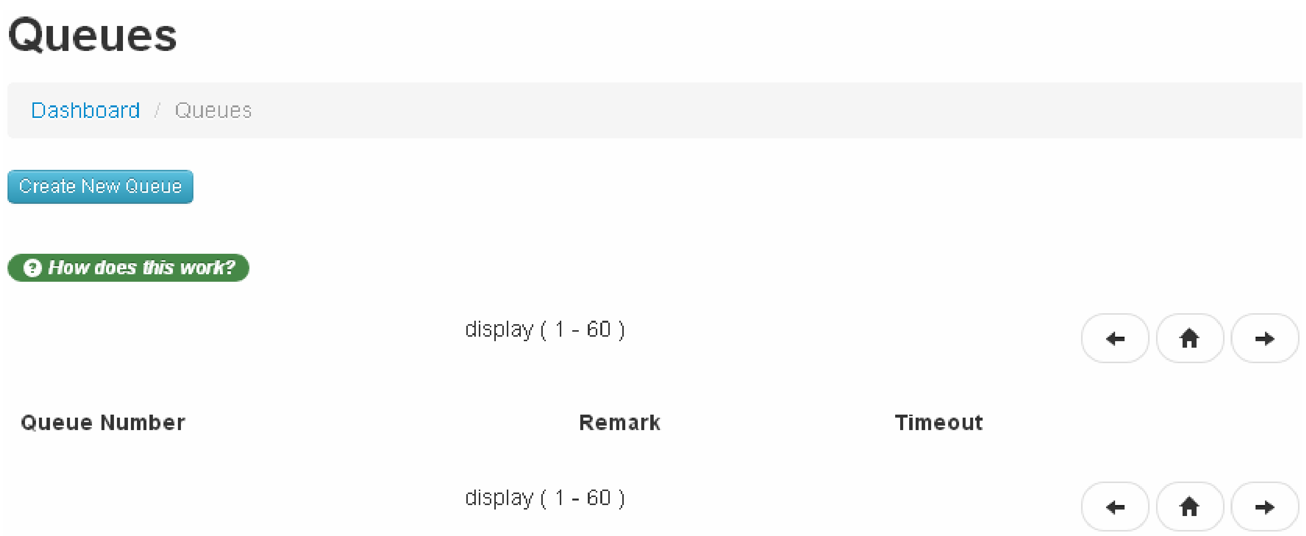




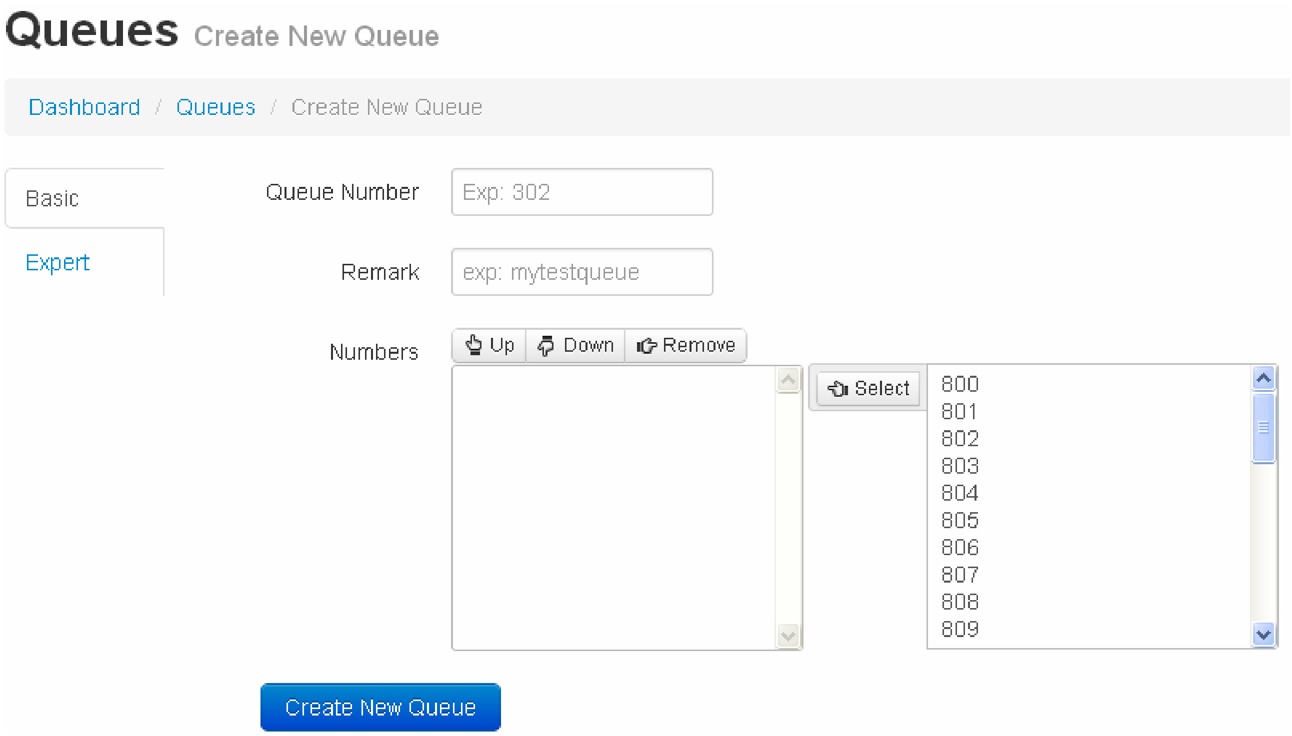
After you set, you can make a call to the conference number via the extension.

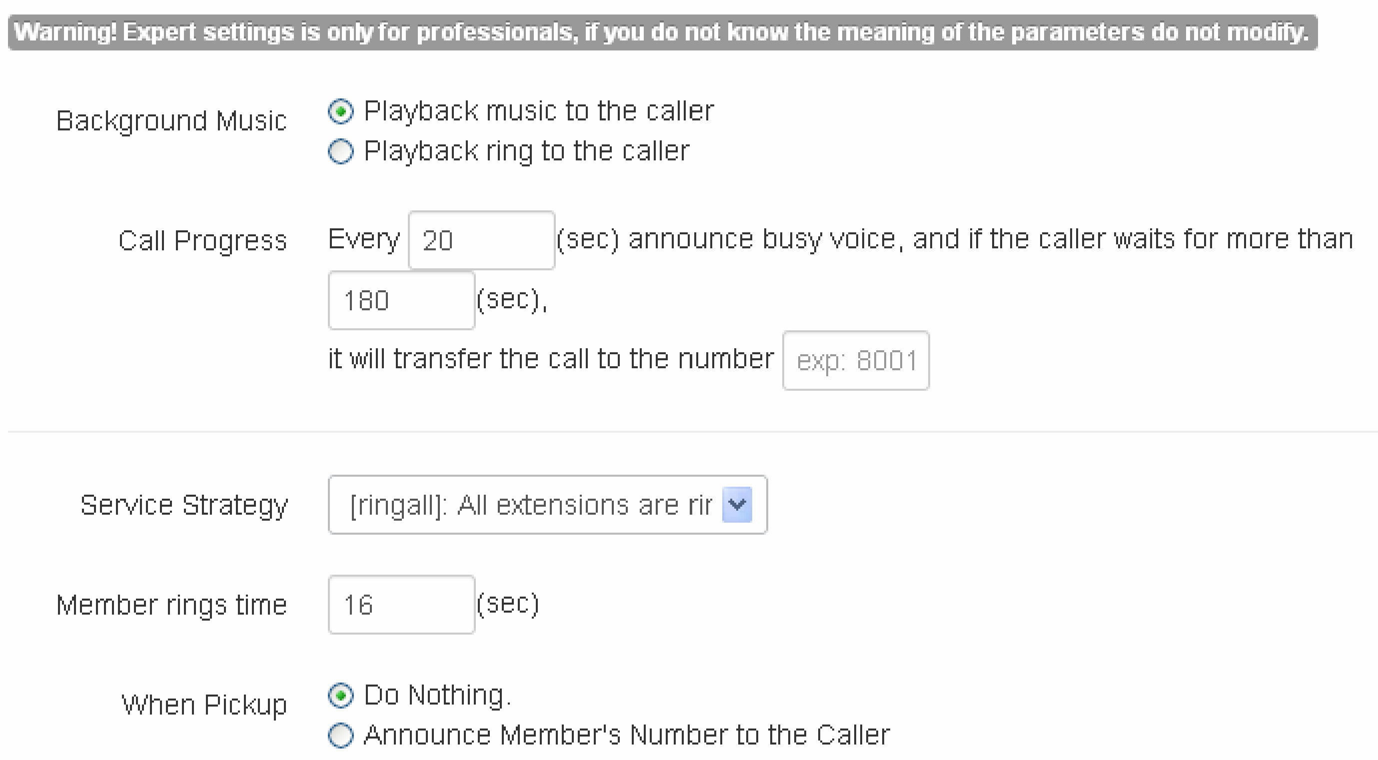
Queues

Select this type if the extension will queue callers to speak with a representative. Call queues are often used to dial into a particular department or group; for example, the extension for the accounting department might be a call queue. You can customize your call queues in the extension setup process.



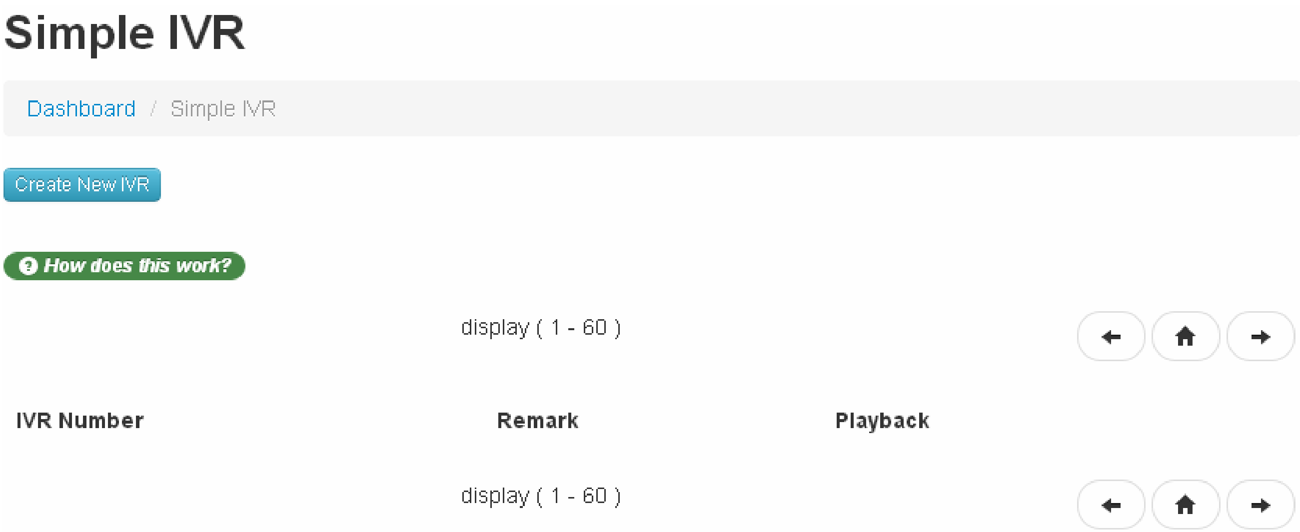
Now for example, we create a Queue, we add 802 and 800 into our queue.



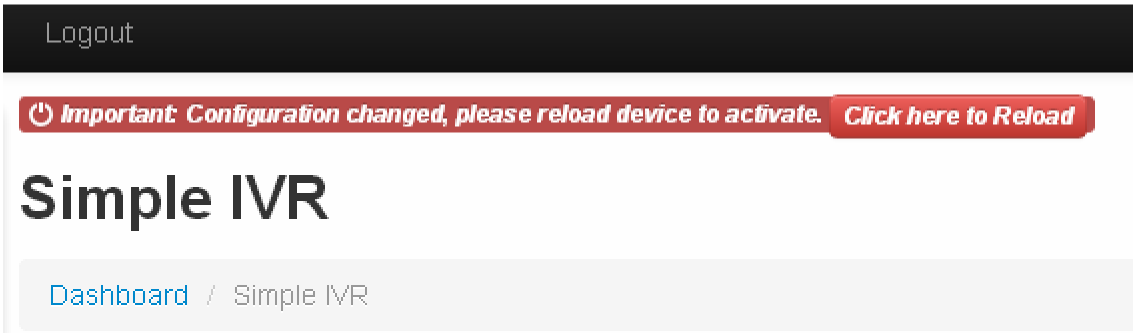


You can also set the service strategy: Ring all, Radom, Memory.

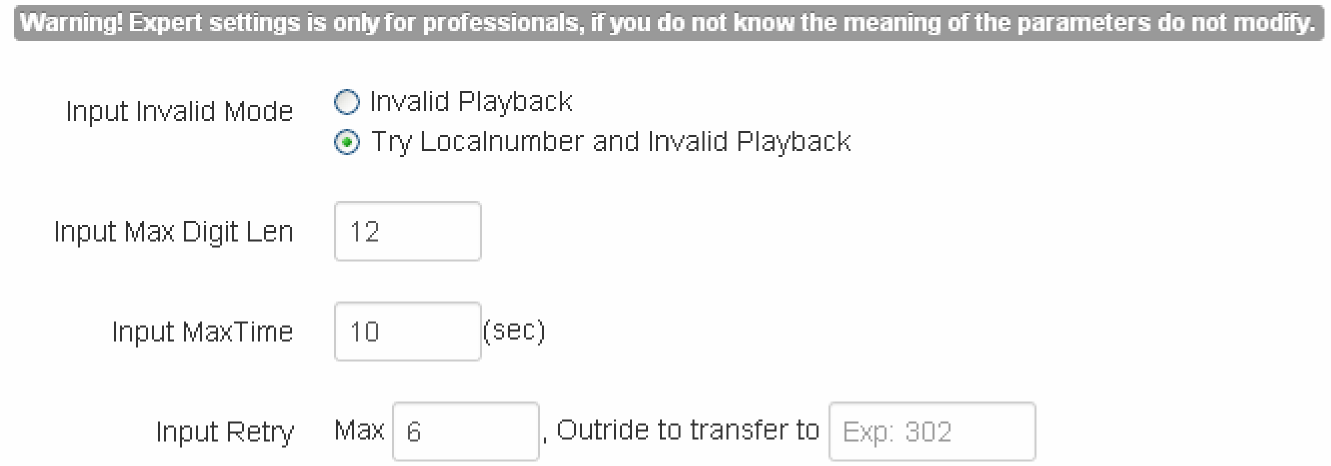
Simple IVR



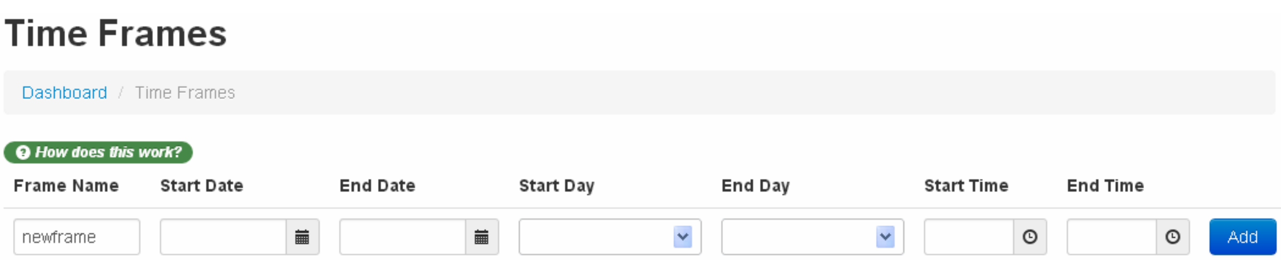
Here we can set the IVR (Voice menu)
For example, we set one IVR number 333, choose the playback file-Welcome, and choose that press 0 to transfer to 800 and press 1 to transfer to 801.Create the new IVR.



Then click the red bar to reload.
Time Frame: You need to set this as below first then come back to choose.
Expert: You can set the mode when user input the invalid number.
Input Max Digit Len: For example, if you input 3 means if you press only one number when you access to the IVR, you need to press that number and # to call directly. Otherwise need to wait the Max Time.
Max Time: the waiting time to connect someone.
Input Retry: For example, if you input continuous two time wrong number, the system will automatically transfer to another number you input.

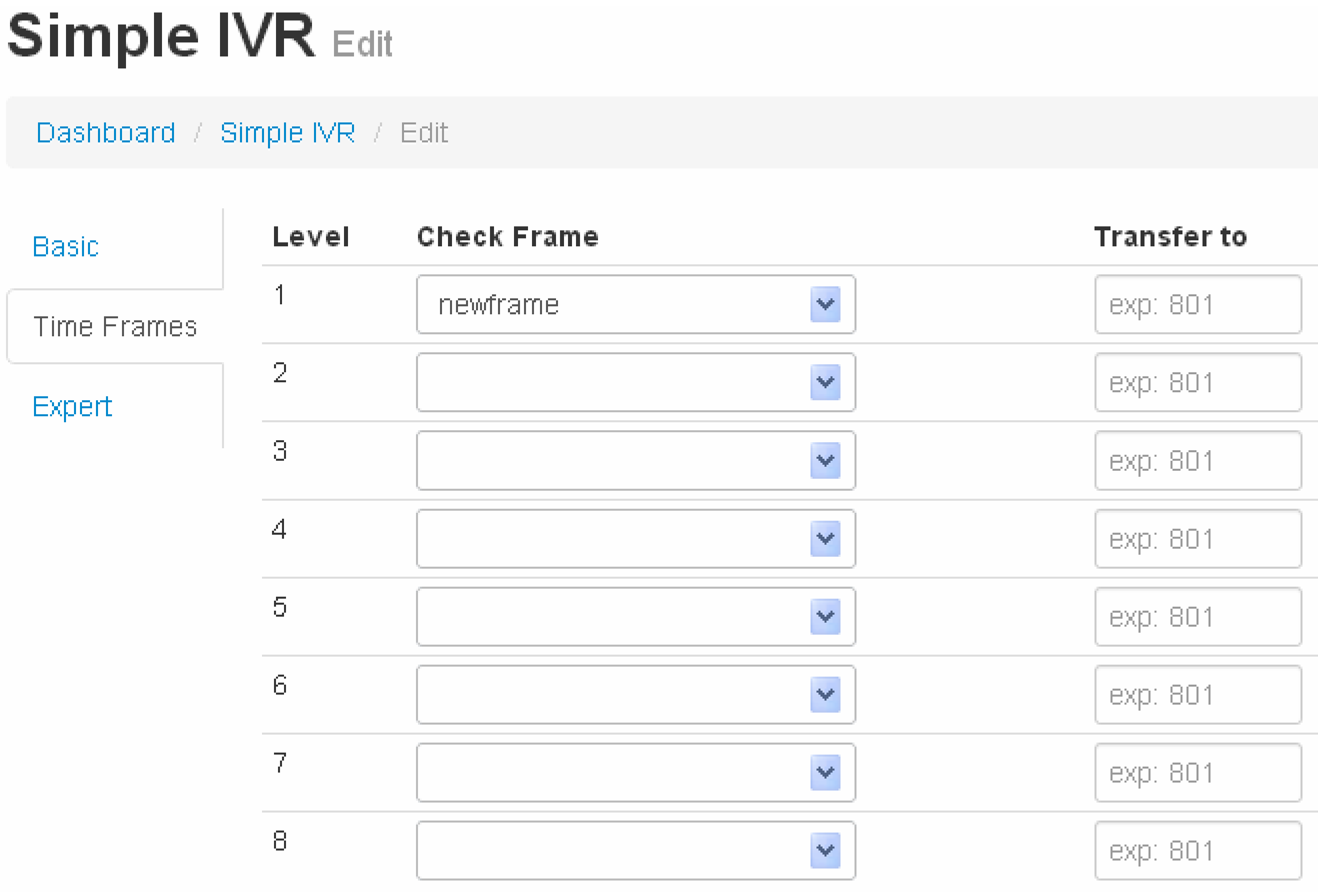


Time Frames

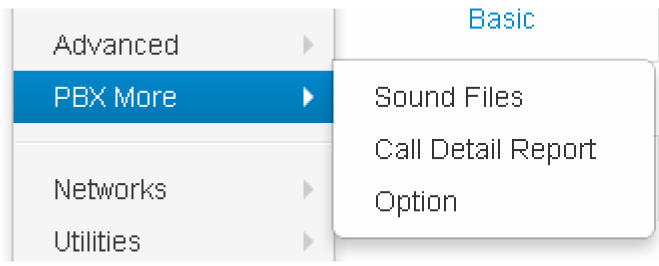


As the above, we create one set working time and click “add”. Now we can come back to the simple IVR- time frame – to set it.
If all the parameters are zero, it means to ignore this time frame.
(Advanced-Simple IVR)

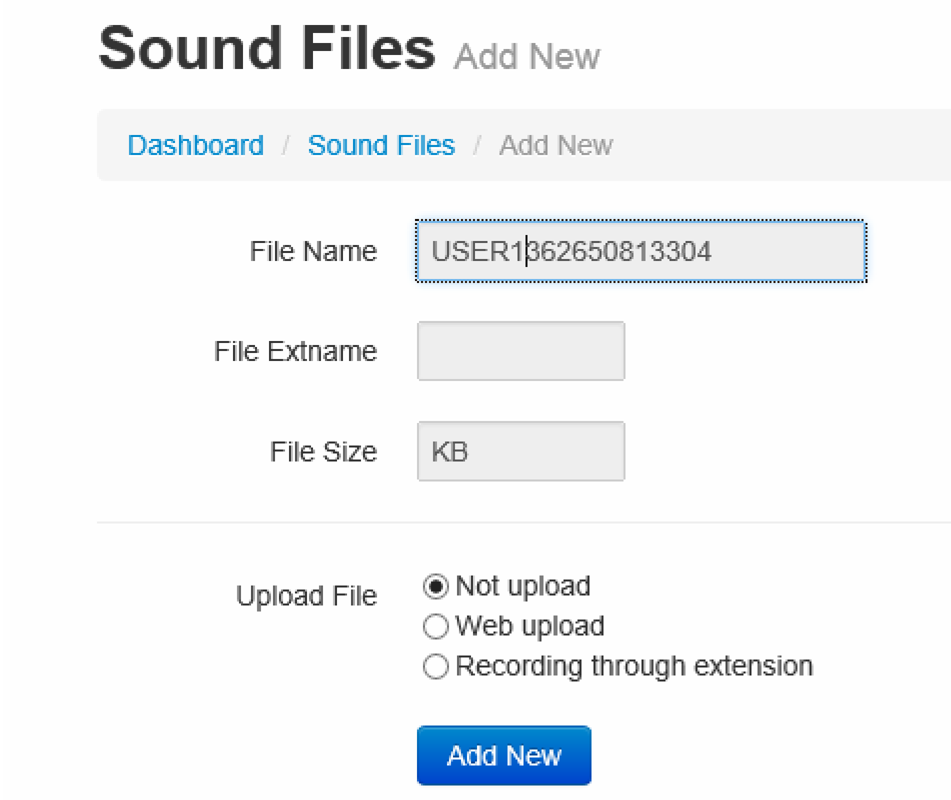
OK. When during the working time you set, all the incoming calls will transfer to 801.



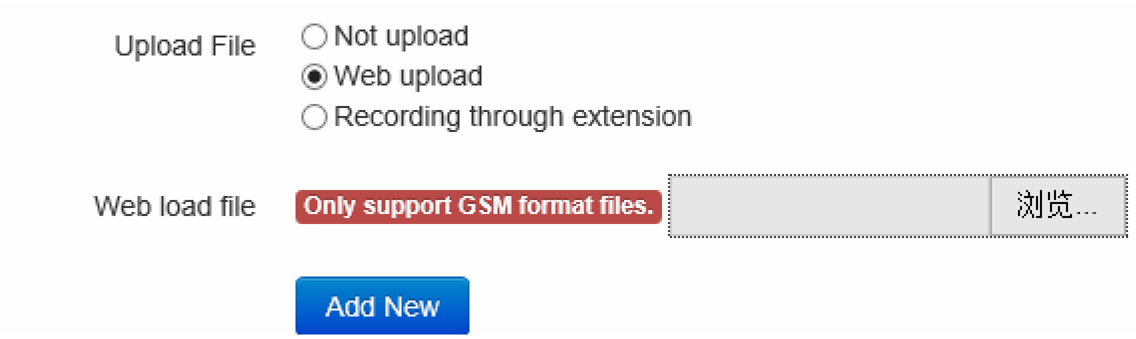
PBX More



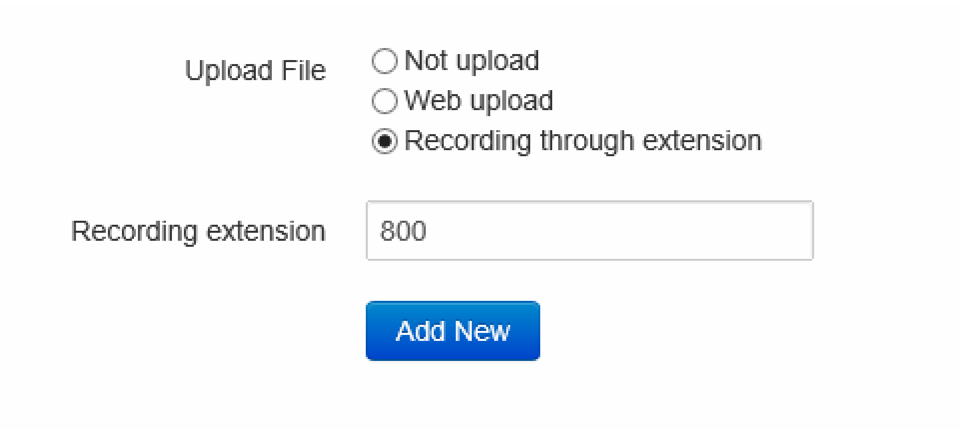
Sound Files



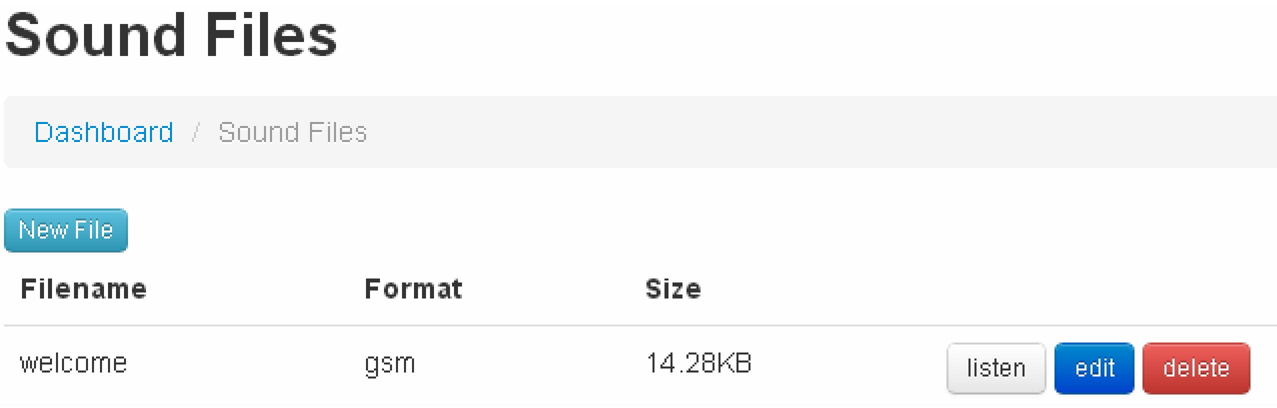
By default, the file name is unmodified when you add a new sound.
It can be used as the Simple IVR's file.
File Ext name/ file size: Unmodified.
Upload file: you can choose the way to upload the file.
Web upload: Only support GSM file.



When you input one extension number, it will record automatically.



When you upload a sound file, you can listen, edit and delete.



Call Detail Report

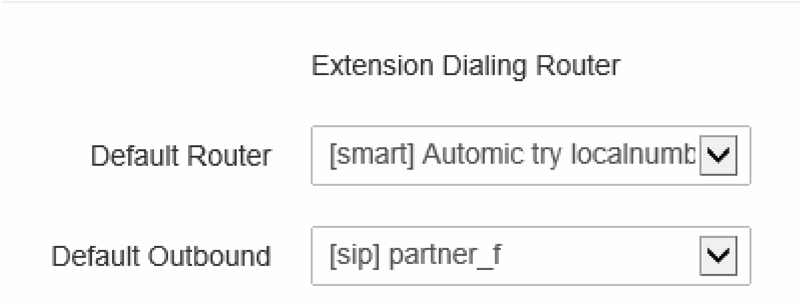
You can check your call detail report here.

Option

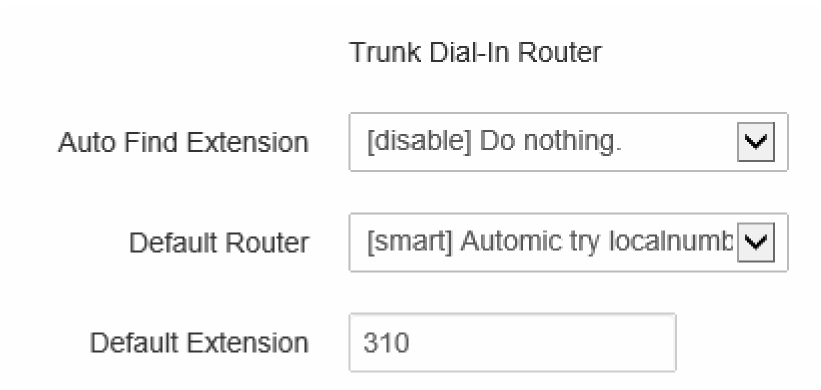
PBX General:

Outbound and internal ring time: you can choose the time.

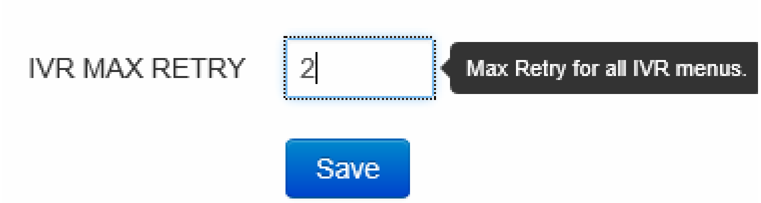
Extension dialing route: you can set the dialing route here as well as the Extension-Dialing Route



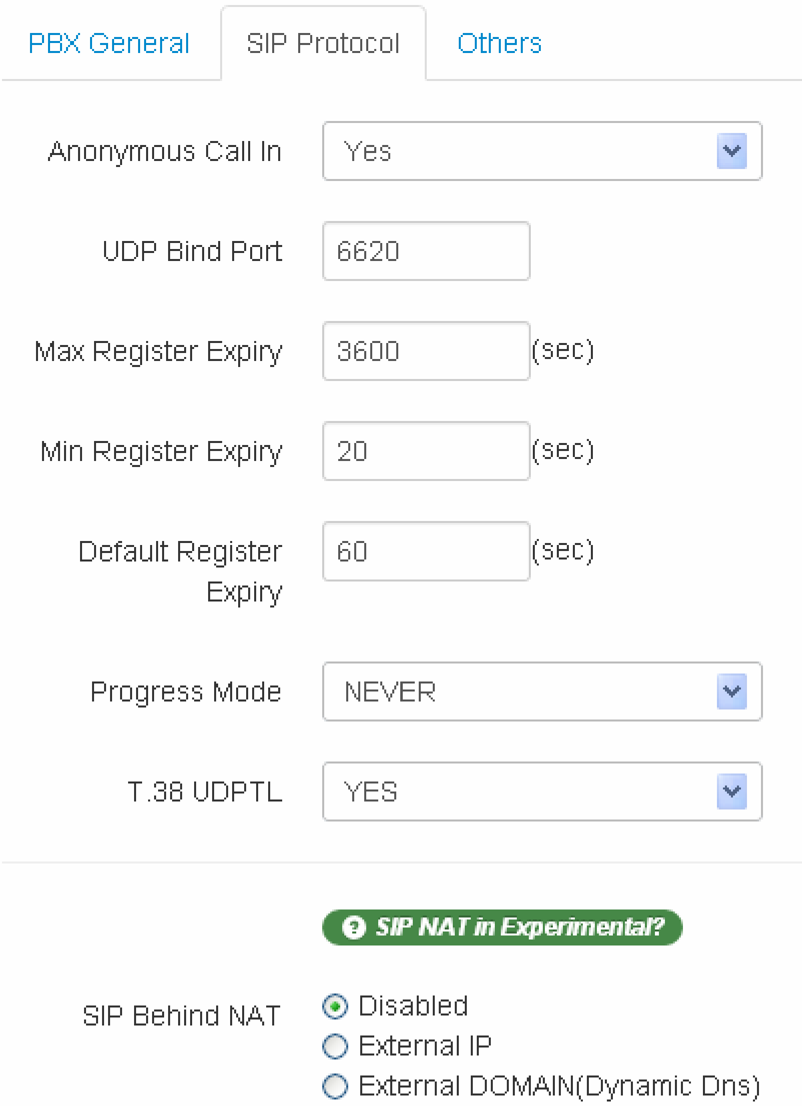
Trunk Dialing route: you can also find the setting in Line provider-Dial-in Route



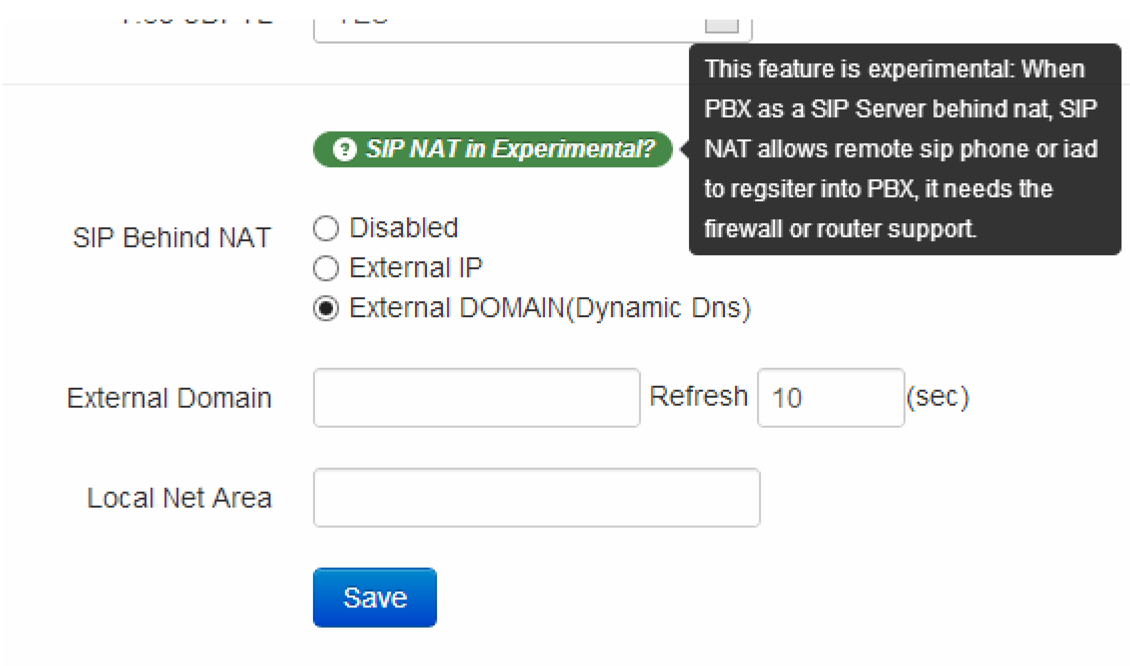
IVR MAX TRY: For example, if we input 2 means the system will automatically quite when the IVR playback for twice.



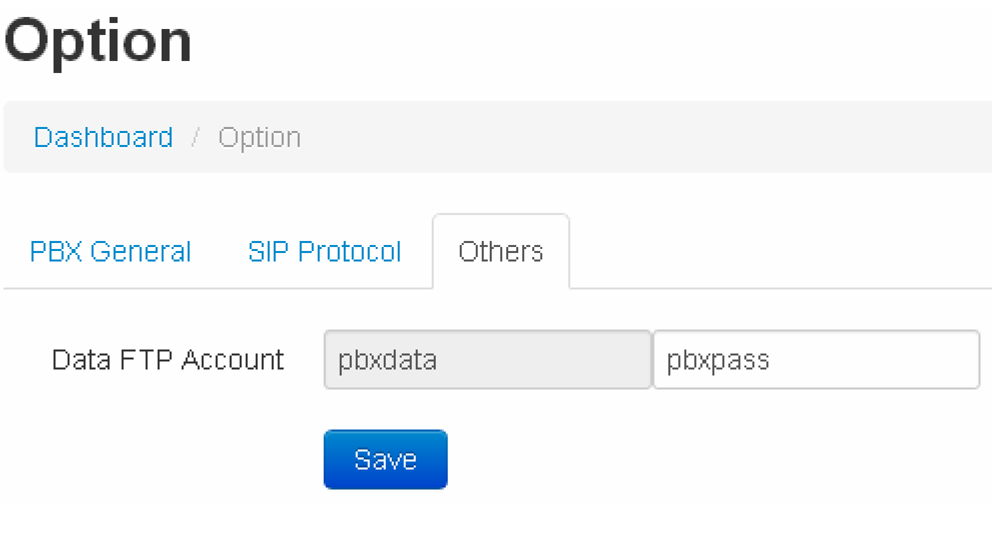
SIP Protocol



Anonymous Call In
UDP Bind Port: by default is 6020
Please note you should modify this port when you set the extension port is 5060.
Max Register Expiry
Min Register Expiry
Default Register Expiry
Progress Mode
T.38 UDPTL
SIP Behind NAT
This experimental feature is that when a PBX as a SIP server behind Nat, SIP Nats allows your SIP phone and IADs to register into PBX. And it needs the firewall and router’s support.



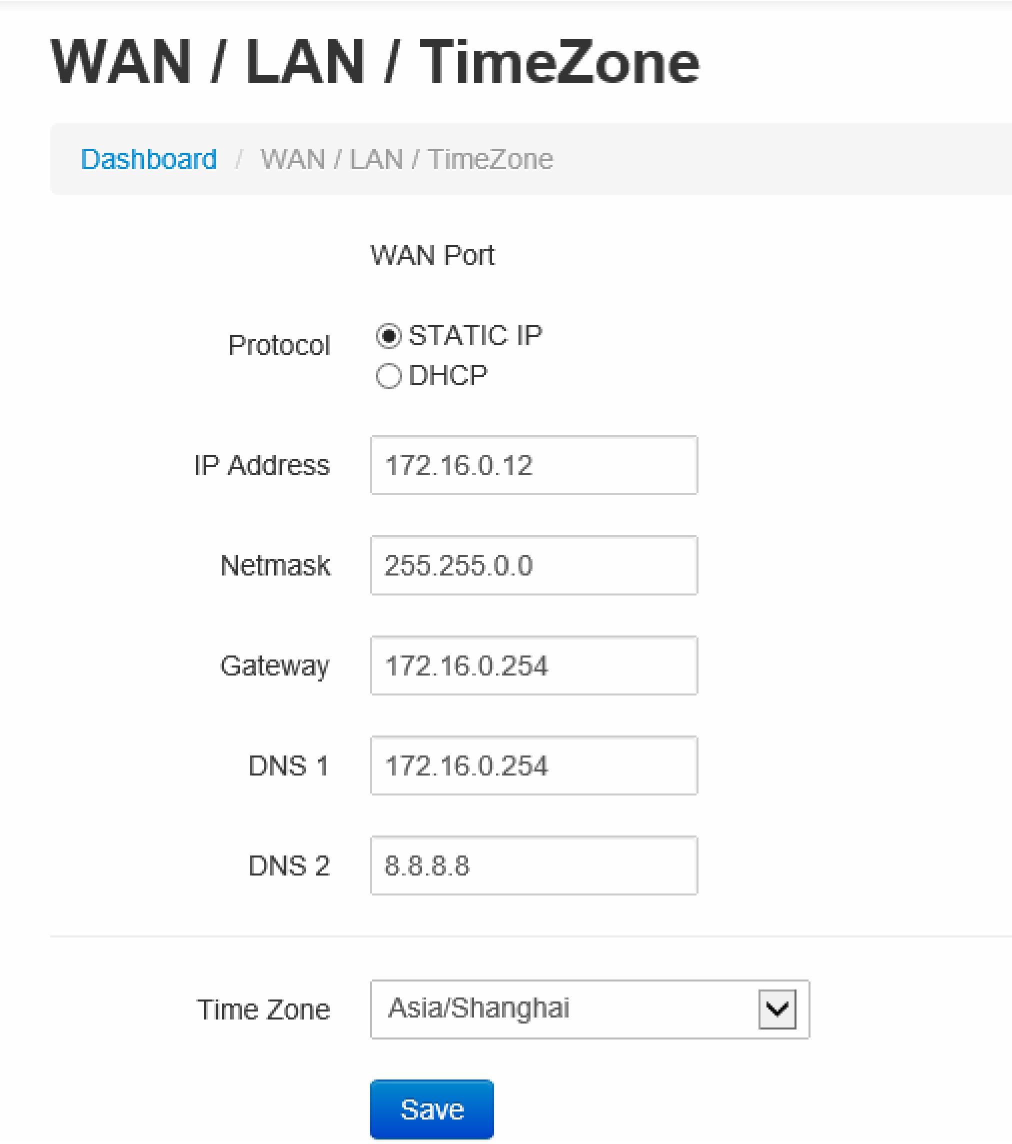
Option
You can check the data FTP account here and modify.



Networks

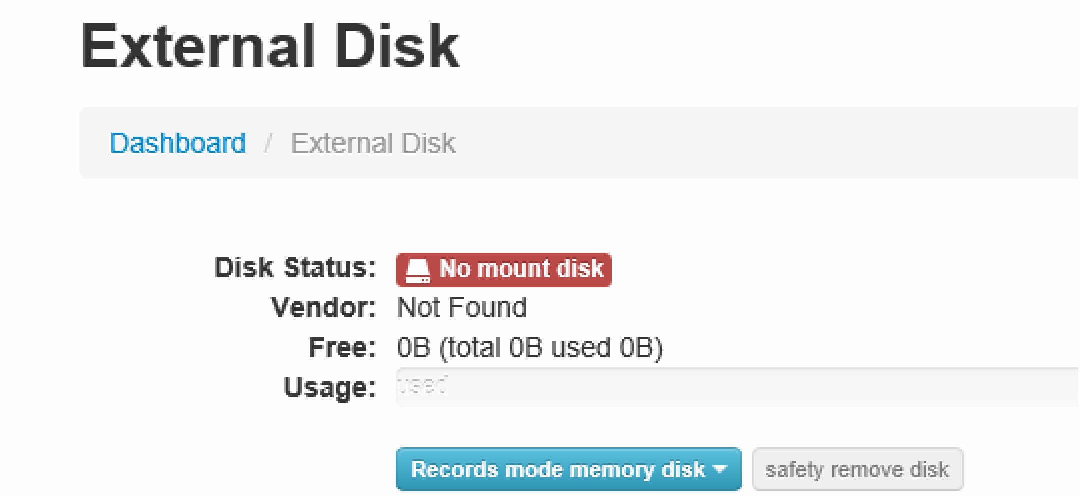
WAN/LAN/Time Zone

You can set it here as well as the Wizard.

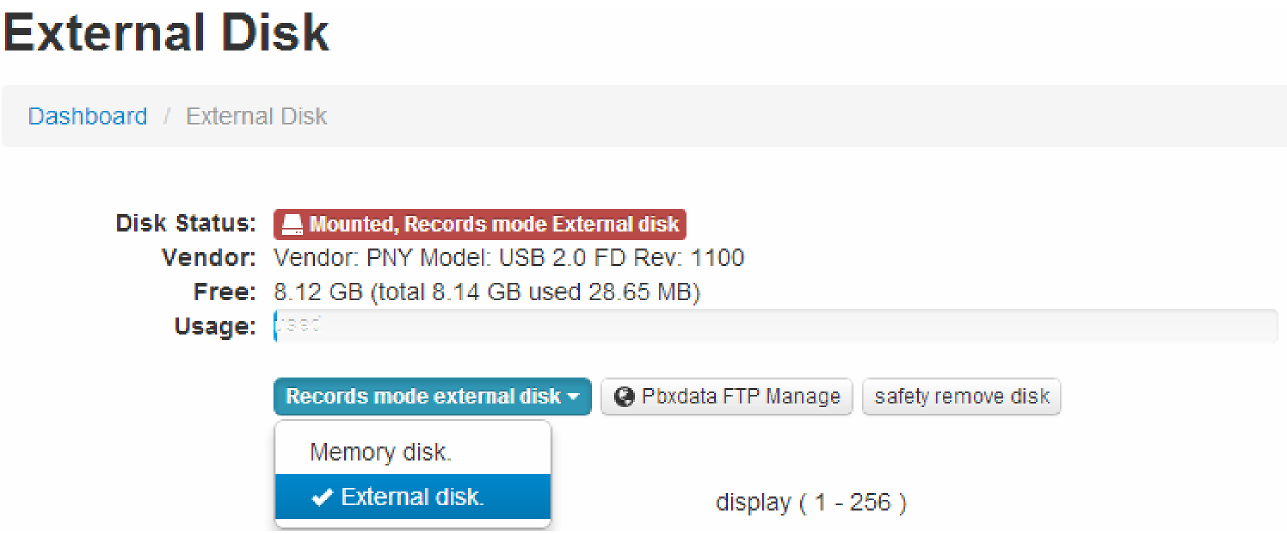


Utilities

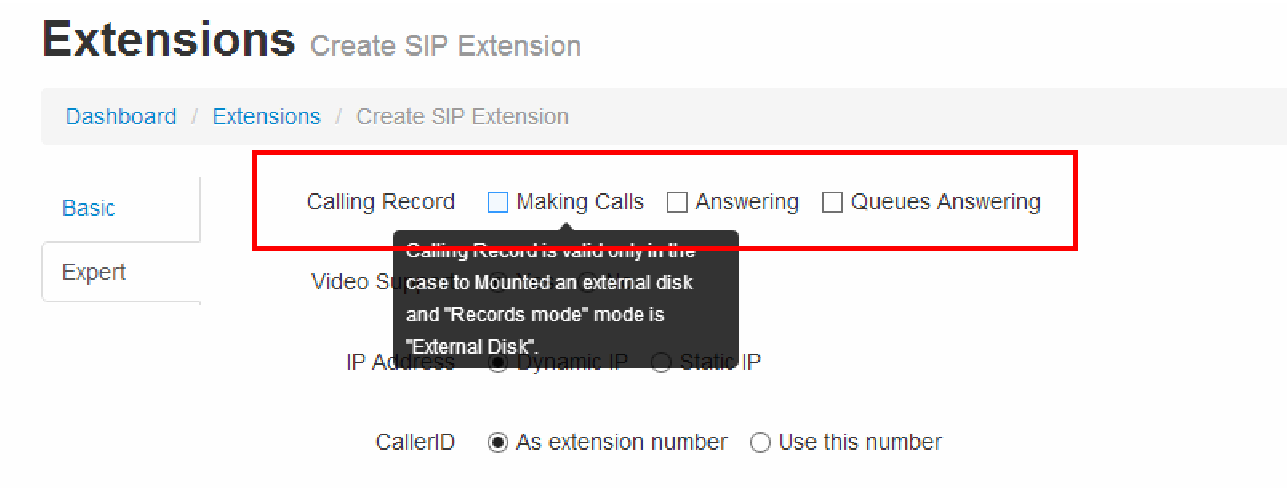
External Disk



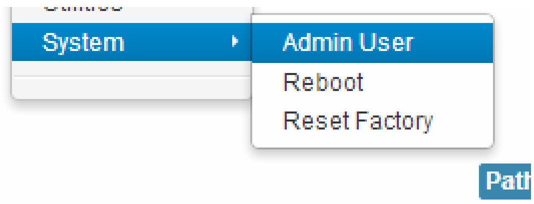
Here you can insert a USB device to record the sound files.
It supports FAT32, EXT4 or based the MLC USB pen disks.
We recommend use the USB device with power adaptor alone.
Insert one USB. Please remember to choose the external disk.



You can also check the PBX data in FTP Manage.
When you want to extract the USB, just click the safety remove disk.
And only in this circumstance, you can find the below menu in the Extensions interface.

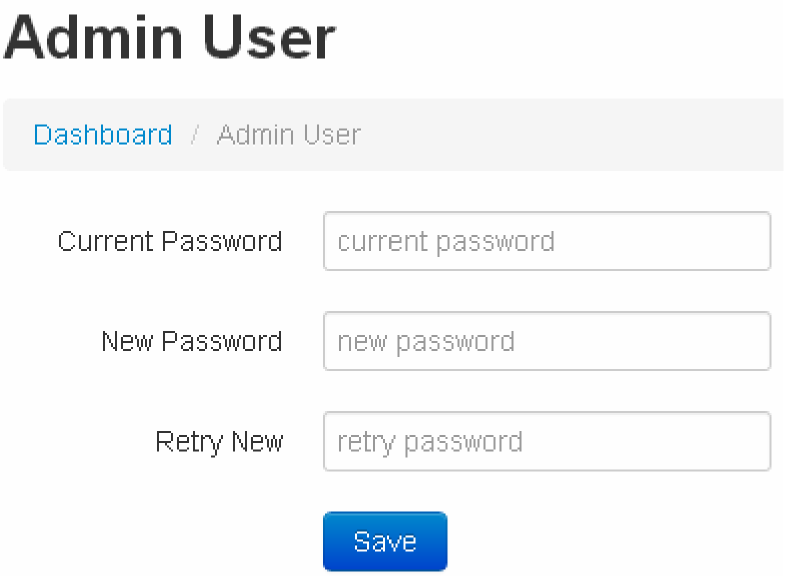


System

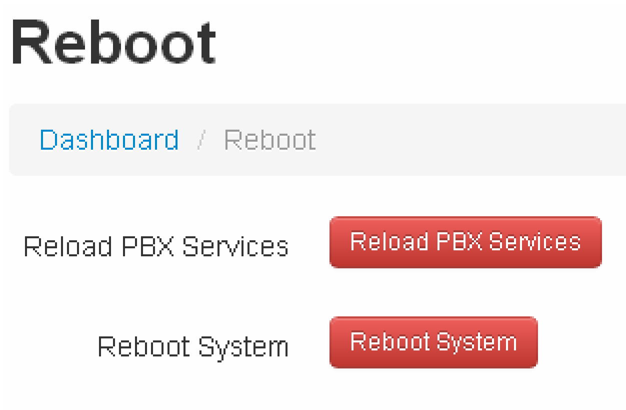


Admin User

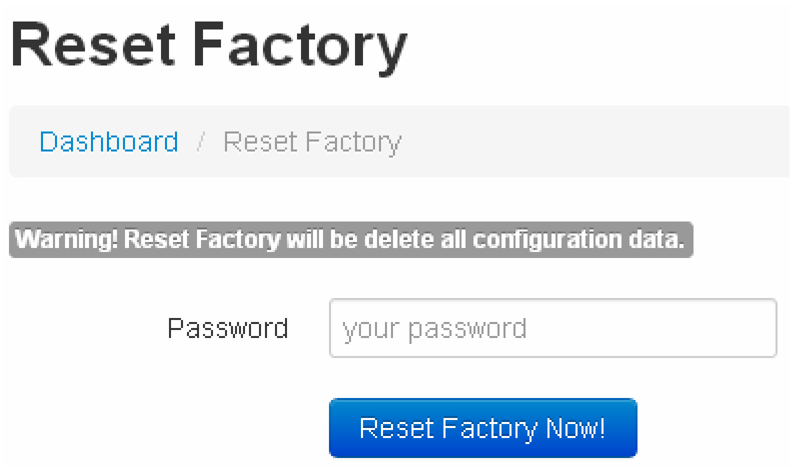
You can set and modify the admin user ,password here.



Reboot



Reset Factory



Factory Mode: Input your password to reset your device.

Glossary

ATA (Analog Telephony Adapter)
A device used to connect one or more standard analog telephones to a digital and/or non-standard telephone system such as a Voice Over IP based network.

DID (Direct Inward Dial)
A feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service, the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk. As calls are presented to the PBX, the dialed destination number (DNIS) is transmitted, usually partially (e.g., last four digits), so that the PBX can route the call directly to the desired telephone extension within the organization without the need for an operator or attendant.
DID numbers are assigned to a communications gateway connected by a trunk to the public switched telephone network (PSTN) and the VoIP network. The gateway routes and translates calls between the two networks for the VoIP user. Calls originating in the VoIP network will appear to users on the PSTN as originating from one of the assigned DID numbers.

DNS (domain name system)
The Internet's name/address resolution service that translates alphabetic domain names into numeric IP addresses. For example, the domain name www.pbx.com might translate to 198.105.232.4. If a computer cannot access DNS, the user's web browser will not be able to find web sites and the user will not be able to receive or send email. The DNS system consists of three components: DNS data,name servers, and Internet protocols for getting the data from the servers.

Domain name server
A computer that runs a program that converts a fully qualified domain name (FQDN) into its numeric
IP address and vice versa.

DTMF (Dual-Tone Multi-Frequency)
The signal that is generated when a user presses the touch keys of an ordinary telephone. Also known as "Touchtone," DTMF has essentially replaced pulse dialling. When a user presses touch keys, two tones of specific frequencies are generated (one from a high-frequency group and the other from a lowfrequency
group), so it's impossible for the voice to imitate the tones.

FTP (File Transfer Protocol)
A standard Internet protocol used to upload and download files between computers that are

connected to the Internet. FTP uses the Internet's TCP/IP protocols as does HTTP, which transfers displayable Web pages and related files, and SMTP, which transfers e-mail.

GSM (Global System for Mobile communication)
A wireless telephone standard in Europe and other parts of the world.GSM uses a variation of time division multiple access (TDMA), which is the most widely used of the three digital wireless telephony technologies (TDMA, GSM, and CDMA). GSM digitizes and compresses data, then sends it down a channel with two other streams of user data, each in its own time slot. It operates at either the 900 MHz or 1800 MHz frequency band.

IP-PBX (Internet Protocol Private Branch Exchange)
A telephone switch (see "PBX") located on a customer's premises that utilize VoIP to manage and deliver calls.

ITSP (Internet Telephone Service Provider)
A company that offers an Internet data service for making telephone calls using VoIP. Most ITSPs use SIP, H.323, or IAX for transmitting telephone calls as IP data packets. Customers may use VoIP phones or traditional telephones with an analog telephony adapter (ATA).

ITU (International Telecommunication Union)
A telecommunications standards body that is guided by the United Nations. It was founded as the International Telegraph Union in Paris on May 17, 1865. The ITU acts as the global focal point for governments and the private sector in developing networks and services and is comprised of more than 185 countries and produces over 200 standards recommendations annually in the areas of information
technology, consumer electronics, broadcasting, and multimedia communications.

IVR (Interactive Voice Response)
A telephone technology that allows a caller to respond to configured voice menus through voice and
touch tone. The IVR system responds with pre-recorded audio to further direct callers on how to proceed.

LAN (Local Area Network)
A computer network covering a small physical area, like a home, office, or small group of buildings, such as a school, or an office park. LANs are connected primarily through Ethernet and can be connected to other LANs over any distance via telephone lines and radio waves. LANs have a high data transfer rate and are not very expensive to set up. See also "WAN."

MAC (Media Access Control) address
A hardware address that uniquely identifies most network adapters or network interface cards (NICs) by the manufacturer for identification. The manufacturer's registered identification number is usually part of the MAC address if it was assigned by the manufacturer. The MAC address is used by the Media Access Control protocol sub-layer of

the Data-Link Layer (DLC) of telecommunication protocols.

MIPS (million instructions per second)
An old method for measuring a computer's speed and power and, by implication, for determining the amount of work a computer can do. It measures the approximate number of machine instructions the computer can execute in 1 second (i.e., it measures CPU speed). Because there are so many variables with computer performance (e.g., varying amounts of time for different instructions, importance of I/O speed, etc.), MIPS ratings are not used that often anymore. However, a MIPS rating can give you a general idea of a computer's speed.

NAT (Network Address Translation or Network Address Translator)
The method for translating an IP address used within one network to a different IP address known within another network (one network is designated the *inside* network and the other is the *outside* network). NAT allows as a router, for example, to act as an agent between the public network (e.g., the Internet) and a private network (i.e., a local network), which means that a single, unique IP address can represent an entire group of computers.

PBX (Private Branch exchange)
A telephone exchange that serves a particular business or office, as opposed to one that is owned by a common carrier or telephone company and is used by many businesses or the general public. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.PBXs have evolved over time, beginning as a manual switchboard or attendant console that was operated by a telephone operator (circuit switching) to the modern IP PBX. See also "IP PBX."

PSTN (Public Switched Telephone Network)
The network of the world's public circuit-switched telephone networks. Originally a network of fixed line analog telephone systems, the PSTN is now almost entirely digital in its core and includes mobile as well as fixed (plain old telephone service, POTS) telephones. The PSTN is largely governed by technical standards created by the ITU-T, and uses E.163/E.164 telephone numbers for addressing.

Proxy Server
A server (a computer system or an application program) that acts as an intermediary for requests from clients seeking resources from other servers. The VoIP proxy server is used in a DMZ of a company's secure internal communication network and receives VoIP control messages and VoIP media streams.
Using the MAC address and source IP address contained in the control message, the proxy server pushes a policy change to the internal network's external firewall to open call control protocol ports and Real Time Protocol (RTP) ports only for packets from the source IP address. The VoIP proxy server hides the company's internal network address and directs incoming VoIP packets to an IP-PBX connected to the company's internal network.

RAM (Random Access Memory)

A form of computer data storage that allows stored data to be accessed in any order (i.e., “random access”).
RAM is used by a computer’s operating system, application programs, and currently used data, so that they can quickly be reached by the computer’s processor. RAM is quickly readable and writeable compared to other kinds of computer storage (e.g., the hard disk, floppy disk, and CD-ROM);However, data in RAM remains only as long as the computer is running. Once the computer has been turned off, RAM loses its data. When the computer is turned on again, the operating system and other files are once again loaded into RAM.

Router
A device for connecting one or more computers to other computers, networked devices, or to other networks. Compared to hubs and switches (which are also connecting types of devices), a router is the smartest and most complicated of the three. Routers can be programmed to understand and route the data its being asked to handle. Configuration is done through a user interface. Larger routers are capable of being programmed to communicate with other routers to determine the best method of getting network traffic from point A to point B. Hubs work at the data link and network layers (layers
2 and 3) of the OSI model.

SIP (Session Initiation Protocol) [RFC 3261, 3262, 3263, 3264, and 3265]
A signalling protocol for initiating and terminating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality (it is used mainly for voice and video calls over the Internet or data networks).

SIP Trunk
A service offered by an ITSP that allows businesses that have a PBX for their internal calls to use VoIP to go outside the enterprise network by using the same connection as the Internet connection. Before SIP trunks can be deployed, a business must have a PBX with a SIP-enabled trunk side, an enterprise edge device that understands SIP, and an ITSP. See “ITSP.”

Soft-switch (software switch)
A term used to describe the software that is used to bridge a public switched telephone network (PSTN) and VoIP. This is done by separating the call control functions of a phone call from the media gateway (transport layer). The soft-switch is typically used to control connections at the junction point between circuit and packet networks.

UDP (User Datagram Protocol) [RFC 768]
A communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that is using the Internet Protocol (IP). UDP merely performs IP traffic demultiplexing based on UDP port numbers, after which it provides a checksum that can be used by end systems to determine whether the datagrams received were corrupted by the network.

WAN (Wide Area Network)
A computer network that covers a broad area (e.g., any network that links across metropolitan, regional, or national boundaries). WANs are similar to the Internet in that they are not owned by a single organization. They exist under collective or distributed ownership and management. For WAN connectivity over the longer distances, ATM, frame relay, and X.25 are used. Computers connected to
a WAN can be connected via the telephone system, leased lines, or satellites. WANs have a lower data transfer rate when compared to LANs. See also “LAN.”